From forrerj@ucs.orst.edu Sun Jun 02 23:12:01 1996
Received: from ucs.orst.edu (root@UCS.ORST.EDU [128.193.4.5]) by tapr.org
(8.7.5/8.7.3/1.9) with SMTP id XAA02444 for <HFSIG@TAPR.ORG>; Sun, 2 Jun 1996
23:11:58 -0500 (CDT)
Received: from p04.t0.monrotel.com by ucs.orst.edu;
(5.65v3.2/1.1.8.2/13Mar96-1233PM)
id AA05223; Sun, 2 Jun 1996 21:11:47 -0700
Message-Id: <1.5.4.16.19960603053332.29172514@ucs.orst.edu>
X-Sender: forrerj@ucs.orst.edu
X-Mailer: Windows Eudora Light Version 1.5.4 (16)
Mime-Version: 1.0
Content-Type: text/plain; charset="us-ascii"
Date: Sun, 02 Jun 1996 21:33:32 -0800
To: HFSIG@tapr.org
From: Johan Forrer <forrerj@ucs.orst.edu>
Subject: QPSK BANDWIDTH ?

Hi all,

Could anyone please comment on the expected bandwidth for a DQPSK signal at
T bauds (or 2*T bits/sec) with raised cosine pulse shaping.

I have been monitoring my QPSK signal using an FFT analyser program. My
modulator uses a root raised cosine response with x/sinx compensation with
alpha rolloff factor 0.2. While the QPSK signal is cycling though all its
possible bit patterns, I observe five peaks in the emitted spectrum as follows:

carrier at   0 dB,
+/- half T at -6 dB,
+-   T at -14 dB.

Everything else is way down in the noise. From this observation, it appears
that the signal is contained into T Herz. Is this what is supposed to be
there? I am a little puzzled by the two components at +/- half T, also not
too clear whether I should or not use the x/sinx compensation in this case
and whether the rolloff factor is suitable. Theoretically, it looks OK on
simulations - The signal sounds nice and clean and melodic, but I may be
missing something.

Any comments and suggestions will be appreciated.

--Johan

From BRYD@KIDD.CO.ZA Mon Jun 03 01:18:30 1996
Received: from igw01 (igw01.kidd.co.za [192.96.246.1]) by tapr.org
(8.7.5/8.7.3/1.9) with SMTP id BAA13457 for <hfsig@tapr.org>; Mon, 3 Jun 1996
01:18:23 -0500 (CDT)
Received: from KIDD.CO.ZA by igw01 with smtp
(Smail3.1.29.1 #3) id m0uQSxs-000P4BC; Mon, 3 Jun 96 08:18 GMT+0200

```
Received: from KenMail-Message_Server by KIDD.CO.ZA
     with Novell_GroupWise; Mon, 03 Jun 1996 08:21:26 +0200
Message-Id: <s1b2a086.016@KIDD.CO.ZA>
X-Mailer: Novell GroupWise 4.1
Date: Mon, 03 Jun 1996 20:16:15 +0200
From: Danie Brynard <BRYD@KIDD.CO.ZA>
To: hfsig@tapr.org
Subject:  [HFSIG:1165] QPSK BANDWIDTH ? -Reply
```

Hi All.

I have made some pic's (jpg format) of my EVM interface. Interesting
persons can email me for copies otherwise I will ask Johan to upload it to
the web page for hf signal processing

73 danie
BRYD@KIDD.CO.ZA

```
From mcdermot@rdxsunhost.aud.alcatel.com  Mon Jun 03 08:17:54 1996
Received: from aud.alcatel.com (rockdal.aud.alcatel.com [128.251.30.1]) by
tapr.org (8.7.5/8.7.3/1.9) with SMTP id IAA24931 for <hfsig@tapr.org>; Mon, 3 Jun
1996 08:17:51 -0500 (CDT)
Received: from rdxsunhost.aud.alcatel.com.Aud.Alcatel.COM by aud.alcatel.com (4.1/
SMI-4.1)
     id AA19420; Mon, 3 Jun 96 08:17:48 CDT
Received: from eagle.aud.alcatel.com by rdxsunhost.aud.alcatel.com.Aud.Alcatel.COM
(4.1/SMI-4.1)
     id AA21315; Mon, 3 Jun 96 08:17:48 CDT
Received: by eagle.aud.alcatel.com (4.1/SMI-4.1)
     id AA02940; Mon, 3 Jun 96 08:17:47 CDT
Date: Mon, 3 Jun 96 08:17:47 CDT
From: mcdermot@rdxsunhost.aud.alcatel.com (Tom Mcdermott)
Message-Id: <9606031317.AA02940@eagle.aud.alcatel.com>
To: hfsig@tapr.org
Subject: Re: [HFSIG:1165] QPSK BANDWIDTH ?
```

```
> Hi all,
>
> Could anyone please comment on the expected bandwidth for a DQPSK signal at
> T bauds (or 2*T bits/sec) with raised cosine pulse shaping.
>
> I have been monitoring my QPSK signal using an FFT analyser program. My
> modulator uses a root raised cosine response with x/sinx compensation with
> alpha rolloff factor 0.2. While the QPSK signal is cycling though all its
> possible bit patterns, I observe five peaks in the emitted spectrum as follows:
>
> carrier at      0 dB,
> +/- half T at  -6 dB,
> +-       T at -14 dB.
>
> Everything else is way down in the noise. From this observation, it appears
> that the signal is contained into T Herz. Is this what is supposed to be
```

> there? I am a little puzzled by the two components at +/- half T, also not
> too clear whether I should or not use the x/sinx compensation in this case
> and whether the rolloff factor is suitable. Theoretically, it looks OK on
> simulations - The signal sounds nice and clean and melodic, but I may be
> missing something.
>
> Any comments and suggestions will be appreciated.
>
> --Johan
>

Johan: it is important that all measurements be done on a linear channel -- ie: a class-C amplifier will significantly modify the spectrum of a DQPSK signal. A staggered-offset-QPSK (OQPSK, or SQPSK) signal will not be widened as badly as straight QPSK, but both will widen when limited or clipped.

Secondly, it is important that a PRBS (of sufficient length) be used to excite the filter, else you may get individual spectral lines showing through. If you have a repetitive pattern, the energy at the discrete spectral lines may confuse things. If your FFT is of short length, and thus it truncates the PRBS pattern, this may also lead to measurement error. Additionally, windowing of the FFT, and compensation of amplitude induced errors of the window may prove useful.

I beleive that x/sinx compensation is correct if you are filtering square-wavish pulses. If you are filtering impulses, then compensation is of course not necessary.

You should have -3 dB at the +/- half T points of the TX output spectrum regardless of the alpha factor, for a root-cosine transfer function that has been properly compensated. It then falls off faster past +/- 1/2T. At +/- T there should be no output (filtered or not). However, it is common to have spectral leakage at +/- 1/T due to feedthrough of the baud clock, and some effort may be needed to track this down. An alpha factor of 0.2 leads to some pretty wild ringing, and it's important to assure the dynamic range of everything is preserved. Also, a sufficient number of filter taps is needed to have good out-of-band rejection at an alpha factor of 0.2, I think about 33 taps were needed for my prototype in Excel to have adequate out of band rejection.

It sounds as though you are making excellent progress, and am glad to see you using root-cosine filters so that the emitted spectrum will be narrow. The root-cosine filter on receive will help assure good noise bandwidth of the receiver. Incidentally, the noise bandwidth of a root-cosine filter is independent of the alpha factor.

- Tom, N5EG

--+-----------------------------

Tom McDermott | "All opinions expressed
Alcatel Network Systems, Inc. | are my own, and do not
mcdermot@aud.alcatel.com | represent those of Alcatel
[ICC'96 Technical Program Secretary] | Network Systems, Inc."
[June 23-27, 1996, Dallas, Tx.   ] |
---------------------------------------+--------------------------

From BRYD@KIDD.CO.ZA Mon Jun 03 08:29:20 1996
Received: from igw01 (igw01.kidd.co.za [192.96.246.1]) by tapr.org
(8.7.5/8.7.3/1.9) with SMTP id IAA25439 for <hfsig@tapr.org>; Mon, 3 Jun 1996
08:28:59 -0500 (CDT)
Received: from KIDD.CO.ZA by igw01 with smtp
(Smail3.1.29.1 #3) id m0uQZgm-000P5cC; Mon, 3 Jun 96 15:28 GMT+0200
Received: from KenMail-Message_Server by KIDD.CO.ZA
with Novell_GroupWise; Mon, 03 Jun 1996 15:32:13 +0200
Message-Id: <s1b3057d.004@KIDD.CO.ZA>
X-Mailer: Novell GroupWise 4.1
Date: Tue, 04 Jun 1996 03:27:36 +0200
From: Danie Brynard <BRYD@KIDD.CO.ZA>
To: hfsig@tapr.org
Subject: Center freq for rtty ?

By the way...

Why was such high tones selected for RTTY ? (2125Hz mark, 2295Hz
space) My hf rig shows much better response at 800 to 1000Hz. Would
it not be better to implement the channel filters for a RTTY DSP modem
around say 800Hz ?

I am busy with a rtty dsp modem for the evm and is now wondering
about some of my earlier choices :-) From a DSP point of view would it
not be better to work around 800Hz ? I have used my PK232 with great
success in the past using my 500Hz CW filter on RTTY.

danie zs6awk

From hardie@duke.usask.ca Mon Jun 03 11:36:58 1996
Received: from duke.usask.ca (duke.usask.ca [128.233.3.13]) by tapr.org
(8.7.5/8.7.3/1.9) with ESMTP id LAA03637 for <hfsig@tapr.org>; Mon, 3 Jun 1996
11:36:52 -0500 (CDT)
Received: from localhost (hardie@localhost) by duke.usask.ca (8.7.3/8.7.3) with
SMTP id KAA32561 for <hfsig@tapr.org>; Mon, 3 Jun 1996 10:36:38 -0600 (CST)
Date: Mon, 3 Jun 1996 10:36:38 -0600 (CST)
From: Pete Hardie <hardie@duke.usask.ca>
To: hfsig@tapr.org
Subject: Re: [HFSIG:1168] Center freq for rtty ?
In-Reply-To: <s1b3057d.004@KIDD.CO.ZA>
Message-ID: <Pine.OSF.3.93.960603102921.3180A-100000@duke.usask.ca>
MIME-Version: 1.0
Content-Type: TEXT/PLAIN; charset=US-ASCII

On Mon, 3 Jun 1996, Danie Brynard wrote:
> Why was such high tones selected for RTTY ? (2125Hz mark, 2295Hz

> space)

Back when I did a little RTTY with my KAM, I found that it could copy RTTY much better if I used the LOwtones command. What this did was that instead of using the 2125Hz Mark with the Space shifted above that, it used the European "low_tones" in which the Space is 1275Hz and then the Mark is shifted above that. I think I also had to use the INVert command.

So perhaps you could implement these low tones.

73 de Pete
ve5va.qrp@usask.ca

From FORRERJ@frl.orst.edu Mon Jun 03 12:28:14 1996
Received: from cornus.FSL.ORST.EDU (root@FSL.ORST.EDU [128.193.112.105]) by tapr.org (8.7.5/8.7.3/1.9) with SMTP id MAA06309 for <hfsig@tapr.org>; Mon, 3 Jun 1996 12:28:10 -0500 (CDT)
Received: from frl.orst.edu (FRL.ORST.EDU [128.193.118.10]) by cornus.FSL.ORST.EDU (8.6.9/8.6.9) with ESMTP id KAA04089 for <hfsig@tapr.org>; Mon, 3 Jun 1996 10:27:59 -0700
Received: from FRL/SpoolDir by frl.orst.edu (Mercury 1.21);
    3 Jun 96 10:28:04 PST8PDT
Received: from SpoolDir by FRL (Mercury 1.21); 3 Jun 96 10:27:38 PST8PDT
From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Organization: Forest Research Lab. Oregon State
To: hfsig@tapr.org
Date: Mon, 3 Jun 1996 10:27:36 -0800
Subject: Re: [HFSIG:1167] Re: QPSK BANDWIDTH ?
Priority: normal
X-mailer: Pegasus Mail v3.22
Message-ID: <58825D8058E@frl.orst.edu>

Hi Tom,

Thanks for the insight - I am learning a lot. Seems like I'm heading in the right direction.

For those that might be wondering what this is about; I have been exploring some of the materials in Tom's soon-to-be-released book on modem design. The section on pulse shaping is particularly good and includes extensive examples that one can play with using an Excel spreadsheet.

The pulses that need to be shaped are from the QPSK I/Q channels prior to modulation and they are indeed approximately square waves - I chose an alpha factor according to what I see being used in V32 modems - but I'll play with some of the other factor to see the effects. In my case the raised cosine filter order is 65 and the FFT program uses 4096 point FFTs that produces some 2 Hz resolution (I do assume that the program applies windows - it's the VE2IQ program).

Pulse shaping has introduced new problems with my clock and carrier

acquisition algorithms, mainly because they are very much based
on signal energy over a baud. Pulse shaping cleans the signal up to
the extent that a lot more work is now needed to obtain and
maintain reliable clock and carrier sync. I probably now need to add a
scrambler to whiten the spectrum. What a lot of fun!

--Johan

From FORRERJ@frl.orst.edu Mon Jun 03 12:35:43 1996
Received: from cornus.FSL.ORST.EDU (root@FSL.ORST.EDU [128.193.112.105]) by
tapr.org (8.7.5/8.7.3/1.9) with SMTP id MAA06670 for <hfsig@tapr.org>; Mon, 3 Jun
1996 12:35:40 -0500 (CDT)
Received: from frl.orst.edu (FRL.ORST.EDU [128.193.118.10]) by cornus.FSL.ORST.EDU
(8.6.9/8.6.9) with ESMTP id KAA04348 for <hfsig@tapr.org>; Mon, 3 Jun 1996
10:35:33 -0700
Received: from FRL/SpoolDir by frl.orst.edu (Mercury 1.21);
    3 Jun 96 10:35:38 PST8PDT
Received: from SpoolDir by FRL (Mercury 1.21); 3 Jun 96 10:35:13 PST8PDT
From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Organization: Forest Research Lab. Oregon State
To: hfsig@tapr.org
Date: Mon, 3 Jun 1996 10:35:12 -0800
Subject: Re: [HFSIG:1168] Center freq for rtty ?
Priority: normal
X-mailer: Pegasus Mail v3.22
Message-ID: <58846303B36@frl.orst.edu>

Hi Danie,

The choice of 2125/2295 tone pair goes back a long way. It however,
probably is not the best choice and you will find that the European standard
specifies 1275/1445.

I dont know of any technical reason, besides IF and IF filters that makes
one better than the other, besides some DSP algorithms that works better
when you have more cycles of the waveform within a signaling element. You
saw a lot of these kinds of technical arguments in the "old days" when
folks used cassette tapes to store computer data.

Not sure this helps.

--Johan

From cbuttsch@slonet.org Mon Jun 03 14:40:02 1996
Received: from spork.callamer.com (cbuttsch@spork.callamer.com [199.74.141.2]) by
tapr.org (8.7.5/8.7.3/1.9) with ESMTP id OAA12364 for <hfsig@tapr.org>; Mon, 3 Jun

1996 14:39:57 -0500 (CDT)
Received: from localhost (cbuttsch@localhost) by spork.callamer.com (8.7.5/8.7.3)
with SMTP id MAA17100; Mon, 3 Jun 1996 12:37:45 -0700 (PDT)
Date: Mon, 3 Jun 1996 12:37:44 -0700 (PDT)
From: Clifford Buttschardt <cbuttsch@slonet.org>
X-Sender: cbuttsch@spork.callamer.com
To: Johan Forrer <FORRERJ@frl.orst.edu>
cc: hfsig@tapr.org
Subject: Re: [HFSIG:1170] Re: QPSK BANDWIDTH ?
In-Reply-To: <58825D8058E@frl.orst.edu>
Message-ID: <Pine.SOL.3.93.960603122853.14881C-100000@spork.callamer.com>
MIME-Version: 1.0
Content-Type: TEXT/PLAIN; charset=US-ASCII

Hi Johan and Tom. I am not sure from your message who Tom might be! In any case, I think that the use of 2125/2975 hertz for rtty eminated from the first use of AFSK on sidband radios. The idea here was to place the tones as high in frequency as possible to avoid any harmonics from being radiated. The lower tones stand a chance that components will fall in the passband. There also were thoughts that the Q of 88 mHy inductors were larger at the higher frequencies. (that's all we had to work with then!)

Let me suggest that you check carefully with Bill DeCarle regarding any programs of his supposedly written for windows. It was our intent to keep things as simple as possible and windows is as far from that concept as could be imagined! Cliff Buttschardt W6HDO/K7RR

On Mon, 3 Jun 1996, Johan Forrer wrote:

> Hi Tom,
>
> Thanks for the insight - I am learning a lot. Seems like I'm heading in the
> right direction.
>
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> exploring some of the materials in Tom's soon-to-be-released book on
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> spreadsheet.
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> modulation and they are indeed approximately square waves - I chose an alpha
> factor according to what I see being used in V32 modems - but I'll play
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> cosine filter order is 65 and the FFT program uses 4096 point FFTs that
> produces some 2 Hz resolution (I do assume that the program applies
> windows - it's the VE2IQ program).
>
> Pulse shaping has introduced new problems with my clock and carrier
> acquisition algorithms, mainly because they are very much based
> on signal energy over a baud. Pulse shaping cleans the signal up to
> the extent that a lot more work is now needed to obtain and
> maintain reliable clock and carrier sync. I probably now need to add a
> scrambler to whiten the spectrum. What a lot of fun!

```
>
> --Johan
>
>
>
>
>
```

```
From FORRERJ@frl.orst.edu Mon Jun 03 15:12:36 1996
Received: from cornus.FSL.ORST.EDU (root@FSL.ORST.EDU [128.193.112.105]) by
tapr.org (8.7.5/8.7.3/1.9) with SMTP id PAA13811 for <hfsig@tapr.org>; Mon, 3 Jun
1996 15:12:33 -0500 (CDT)
Received: from frl.orst.edu (FRL.ORST.EDU [128.193.118.10]) by cornus.FSL.ORST.EDU
(8.6.9/8.6.9) with ESMTP id NAA09281 for <hfsig@tapr.org>; Mon, 3 Jun 1996
13:12:30 -0700
Received: from FRL/SpoolDir by frl.orst.edu (Mercury 1.21);
    3 Jun 96 13:12:35 PST8PDT
Received: from SpoolDir by FRL (Mercury 1.21); 3 Jun 96 13:12:30 PST8PDT
From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Organization:  Forest Research Lab. Oregon State
To: hfsig@tapr.org
Date:          Mon, 3 Jun 1996 13:12:27 -0800
Subject:       Re: [HFSIG:1170] Re: QPSK BANDWIDTH ?
Priority: normal
X-mailer: Pegasus Mail v3.22
Message-ID: <58AE5537E04@frl.orst.edu>
```

Hi Cliff,

Thanks for the note on the high tones - I also built those 88mH
demodulators way back - had a lot of fun!

About "windows" - I trust you mean windows like in Hamming or Blackman-
Harris, not Microsoft? I use Bill's program under DOS and I suspect the
anti-alias filter a little, otherwise, the program works reasonably well,
just don't know whether he actually use a window, i.e., a raised cosine or
such prior to the FFT. If not, its not a matter of life and death, just the
spectrum is a little smeared due to Gibbs affects. That is what Tom
McDermot, N5EG, was hinting at. He has written an excellent book on modems
for the radio communications market, and I have had the pleasure to be a
reviewer - the book will be out soon.

Trust you doing OK? Hows the plans for moving coming along?

73's

--Johan

```
From karn@qualcomm.com Mon Jun 03 22:17:48 1996
Received: from servo.qualcomm.com (servo.qualcomm.com [129.46.128.14]) by tapr.org
(8.7.5/8.7.3/1.9) with ESMTP id WAA03345 for <hfsig@tapr.org>; Mon, 3 Jun 1996
22:17:44 -0500 (CDT)
Received: (from karn@localhost) by servo.qualcomm.com (8.7.5/1.0/8.7.2/1.8) id
```

UAA00621; Mon, 3 Jun 1996 20:17:05 -0700 (PDT)
Date: Mon, 3 Jun 1996 20:17:05 -0700 (PDT)
From: Phil Karn <karn@qualcomm.com>
Message-Id: <199606040317.UAA00621@servo.qualcomm.com>
To: hfsig@tapr.org
In-reply-to: <1.5.4.16.19960603053332.29172514@ucs.orst.edu> (message from Johan Forrer on Sun, 2 Jun 1996 23:15:01 -0500 (CDT))
Subject: Re: [HFSIG:1165] QPSK BANDWIDTH ?

Johan,

Are you really testing with random data? From your description and from your results, it sounds like you're using repeating patterns.

You need to use random data and average your results over a fairly long interval.

Phil

From karn@qualcomm.com Mon Jun 03 22:24:12 1996
Received: from servo.qualcomm.com (servo.qualcomm.com [129.46.128.14]) by tapr.org (8.7.5/8.7.3/1.9) with ESMTP id WAA03771 for <hfsig@tapr.org>; Mon, 3 Jun 1996 22:24:08 -0500 (CDT)
Received: (from karn@localhost) by servo.qualcomm.com (8.7.5/1.0/8.7.2/1.8) id UAA00633; Mon, 3 Jun 1996 20:23:33 -0700 (PDT)
Date: Mon, 3 Jun 1996 20:23:33 -0700 (PDT)
From: Phil Karn <karn@qualcomm.com>
Message-Id: <199606040323.UAA00633@servo.qualcomm.com>
To: hfsig@tapr.org
In-reply-to: <s1b3057d.004@KIDD.CO.ZA> (message from Danie Brynard on Mon, 3 Jun 1996 08:30:45 -0500 (CDT))
Subject: Re: [HFSIG:1168] Center freq for rtty ?

>Why was such high tones selected for RTTY ? (2125Hz mark, 2295Hz
>space) My hf rig shows much better response at 800 to 1000Hz. Would
>it not be better to implement the channel filters for a RTTY DSP modem
>around say 800Hz ?

I suppose it depends on the radio. You should operate near the center of whatever passband you have to avoid the phase distortion that is invariably associated with sharp filters near cutoff.

Phil

From karn@qualcomm.com Mon Jun 03 22:29:07 1996
Received: from servo.qualcomm.com (servo.qualcomm.com [129.46.128.14]) by tapr.org (8.7.5/8.7.3/1.9) with ESMTP id WAA03871 for <hfsig@tapr.org>; Mon, 3 Jun 1996 22:29:03 -0500 (CDT)
Received: (from karn@localhost) by servo.qualcomm.com (8.7.5/1.0/8.7.2/1.8) id UAA00648; Mon, 3 Jun 1996 20:28:26 -0700 (PDT)
Date: Mon, 3 Jun 1996 20:28:26 -0700 (PDT)
From: Phil Karn <karn@qualcomm.com>
Message-Id: <199606040328.UAA00648@servo.qualcomm.com>

To: mcdermot@rdxsunhost.aud.alcatel.com (Tom Mcdermott)
CC: hfsig@tapr.org
In-reply-to: <9606031317.AA02940@eagle.aud.alcatel.com>
(mcdermot@rdxsunhost.aud.alcatel.com)
Subject: Re: [HFSIG:1167] Re: QPSK BANDWIDTH ?

Tom,

Speaking of root raised cosine filters, would you be willing to design a few for me for use in my SQPSK modem? I have the code for your book, but I don't have Excel installed. I have the classic Parkes-McClellan FORTRAN code, but it doesn't seem to be readily adaptable to an arbitrary amplitude response.

I've been rewriting my modem code to use an arbitrary length FIR filter. At the moment I've kept the original 8-sample boxcar shape that was inherent in my earlier explicit integrate-and-dump implementation.

I'm using impulse signalling, so the transmit and receive filters can be the same. Give me a alpha of about 0.5 or so (enough to keep the energy at 1200 baud well within a typical SSB bandwidth). The baud rate is 1200, the sampling rate is 9600. Will a 33 tap filter be sufficient?

Phil

From karn@qualcomm.com Mon Jun 03 22:47:51 1996
Received: from servo.qualcomm.com (servo.qualcomm.com [129.46.128.14]) by tapr.org
(8.7.5/8.7.3/1.9) with ESMTP id WAA04761 for <hfsig@tapr.org>; Mon, 3 Jun 1996
22:47:48 -0500 (CDT)
Received: (from karn@localhost) by servo.qualcomm.com (8.7.5/1.0/8.7.2/1.8) id
UAA00709; Mon, 3 Jun 1996 20:47:16 -0700 (PDT)
Date: Mon, 3 Jun 1996 20:47:16 -0700 (PDT)
From: Phil Karn <karn@qualcomm.com>
Message-Id: <199606040347.UAA00709@servo.qualcomm.com>
To: hfsig@tapr.org
In-reply-to: <58825D8058E@frl.orst.edu> (FORRERJ@frl.orst.edu)
Subject: Re: [HFSIG:1170] Re: QPSK BANDWIDTH ?

Johan,

You might consider the approach I've taken for symbol timing in my satellite SQPSK (aka OQPSK) modem. Rather than extract symbol timing from the data, I do it in a one-shot fashion at the beginning of every packet with a modified Barker sync sequence. Since this is a packet modem with a bounded packet length, I know that my one-shot estimate should hold as long as the crystal clocks on the two ends are reasonably accurate.

For carrier recovery, I also start with a one-shot estimate taken from a carrier preamble, but because carrier frequency is more uncertain I use the preamble estimate to seed an a posteriori 4-phase Costas loop that updates the frequency from block to block.

The Costas loop used to work on raw symbols, but I just changed my code so that once symbol timing is established, the incoming samples are downconverted to DC, LP filtered, rate converted and complex sampled at 2x the baud rate using a fixed LO downconversion frequency derived from the carrier burst in the preamble.

Then the Costas loop only has to correct for the slow phase rotation ("slow" relative to the data rate) caused by any residual error in the preamble frequency estimation (plus any doppler). Although complex samples take more operations to process, I figure I probably save overall from the rate conversion from 9600 samples/sec to 2400. And I've already LP filtered, so I don't have to do that again.

I only need 2x sampling because of the I/Q symbol staggering; true QPSK would only require 1x sampling once symbol timing is established.

Phil

From karn@qualcomm.com Mon Jun 03 23:04:14 1996
Received: from servo.qualcomm.com (servo.qualcomm.com [129.46.128.14]) by tapr.org (8.7.5/8.7.3/1.9) with ESMTP id XAA06066 for <hfsig@tapr.org>; Mon, 3 Jun 1996 23:04:11 -0500 (CDT)
Received: (from karn@localhost) by servo.qualcomm.com (8.7.5/1.0/8.7.2/1.8) id VAA00740; Mon, 3 Jun 1996 21:03:34 -0700 (PDT)
Date: Mon, 3 Jun 1996 21:03:34 -0700 (PDT)
From: Phil Karn <karn@qualcomm.com>
Message-Id: <199606040403.VAA00740@servo.qualcomm.com>
To: hfsig@tapr.org
In-reply-to: <58846303B36@frl.orst.edu> (FORRERJ@frl.orst.edu)
Subject: Re: [HFSIG:1171] Re: Center freq for rtty ?

>I dont know of any technical reason, besides IF and IF filters that makes
>one better than the other, besides some DSP algorithms that works better
>when you have more cycles of the waveform within a signaling element. You

I've given this some thought recently, since I noticed some interesting artifacts in my SQPSK modem receive constellation. It's actually inherent in modulating a low frequency carrier with boxcar pulses.

I had chosen a carrier frequency of 1600 Hz and a baud rate of 1200, so there isn't an integral number of carrier cycles per symbol. So when I did a boxcar integrate-and-dump, the I and Q channels aren't perfectly orthogonal even when you have the carrier phase right. I believe this was contributing to about a 1dB implementation loss in my modem.

In the frequency domain, this effect shows up as the lower spectral sidelobes (for unfiltered BPSK) "wrapping around" DC and aliasing back up into the signal spectrum.

I verified my suspicions by testing at a carrier frequency of 2400 Hz, exactly 2x the baud rate. It got rid of these spectral artifacts by overlaying them in "safe" locations (or in the time domain, it ensured an integral number of carrier cycles per symbol integration time.)

But this is not a practical solution for two reasons. First, you can't guarantee that the receiver will get an incoming carrier at exactly 2400 bps, and secondly most SSB radios won't work that high anyway (the first null above the carrier would occur at 3600 Hz, well outside most SSB filters). I chose 1600 Hz in the first place precisely to center the signal spectrum in a typical SSB filter bandwidth.

So I've been rewriting my modem code to include FIR filters in the I and Q channels in both the transmit and receive data paths. With the appropriate filter response, the sidelobes should be low enough to prevent spectral wraparound and aliasing. Or in the time domain, the filter impulse should be small enough at the beginning and end of the symbol time (where the carrier phase transition takes place) to minimize the effect of the fractional carrier cycle in the sampled filter output.

Of course, another fix would be even "sweeter" -- using two product detectors in quadrature in the receiver and feeding complex audio samples to the modem. This would allow the modem to distinguish easily between positive and negative frequency components. Unfortunately, most SSB transceivers don't have quadrature audio outputs...

Phil

From silbaugh@apci.net Mon Jun 03 23:27:33 1996
Received: from hilly.apci.net (root@hilly.apci.net [206.100.36.3]) by tapr.org (8.7.5/8.7.3/1.9) with SMTP id XAA06886 for <HFSIG@tapr.org>; Mon, 3 Jun 1996 23:27:30 -0500 (CDT)
Received: from dialup178.apci.net (dialup178.apci.net [206.100.36.178]) by hilly.apci.net (8.6.12/8.6.9) with SMTP id XAA22147; Mon, 3 Jun 1996 23:28:13 -0500
Date: Mon, 3 Jun 1996 23:28:13 -0500
Message-Id: <199606040428.XAA22147@hilly.apci.net>
X-Sender: silbaugh@apci.net
X-Mailer: Windows Eudora Light Version 1.5.2
Mime-Version: 1.0
Content-Type: text/plain; charset="us-ascii"
To: Johan Forrer <forrerj@ucs.orst.edu>
From: "Eric E. Silbaugh" <silbaugh@apci.net>
Subject: Re: QPSK BANDWIDTH ?
Cc: HFSIG@tapr.org

Johan,

First let's ignore the differential encoding and raised cosine filetering. Straight QPSK at T bauds/sec should

have the classic sin(x)/x squared power spectrum with a main lobe width (null-to-null) of T Hz. What you consider the 'occupied' bandwidth of this signal depends on your definition; for now let's use the null-to-null bandwidth.

Now add raised cosine filtering. The frequency response of the RC filter (or composite of the RX and TX sqrt RC filters) will multiply the power spectrum of the original QPSK waveform to provide the resulting power spectrum. With a 0.2 rolloff, the ideal raised cosine spectrum will have a bandwidth of 1.2T Hz. The percent excess bandwidth is equal to the rolloff factor.

In the real world your filter can only approximate the ideal RC impulse response. Thus, the spectrum may not be identically zero beyond 0.6T Hz away from the carrier frequency. If your filter is a good approximation there should be little 'splatter' beyond 0.6T Hz.

The previous discussion assumed the data symbols were uncorrelated. Differential encoding adds some memory; which means your symbols may not be uncorrelated anymore. The power spectrum of a digitally modulated signal depends on the autocorrelation of the data. (see J. G. Proakis, M. Salehi 'Communication Systems Engineering,' p.537, Prentice-Hall, 1994) I think the effects of differential encoding on the spectrum should be minimal. It certainly should not change the major features of the spectrum.

Sin(x)/x compensation effects should also be minimal, as long as your sampling rate is well above the Nyquist rate. Again, compensation should not change the major features of the spectrum.

All this is a very long way of saying a bandwidth of about 1.2T Hz (almost T Hz) should be expected for your RC filtered DQPSK signal of T baud/sec.

I cannot explain the discrete components you noted. How do you create the signal you feed into the FFT? How many baud periods do you use in the FFT'd signal? You will not get a good estimate of the average power spectrum from signals only a few baud long.

To estimate the power spectrum with more detail I would create a signal from several tens or hundreds of random bits (independent) and FFT this long signal. Yes this will require a long FFT, but it's just like turning down the resolution bandwidth of a spectrum analyzer. The spectral resolution of the FFT is inversely proportional to the time duration of the signal you transform (longer

times mean finer resolution).

To estimate the average power spectrum you will probably want to use the same signal and compute the periodogram. Just average the FFTs of shorter segments of the signal. The FFT segments can overlap, but should probably be an integer number of bauds in length. You will be trading off resolution for averaging. See any good DSP book for details.

No clue as to the suitability of your choosen rolloff factor. If it works I guess it's good enough! A factor of 0.2 seems to be a reasonable compromize from a bandwidth standpoint. How much amplitude fluctuation in the time-domain does the filtering add?

Hope this helps, sorry for the bandwidth!

Eric, N2NNP

From forrerj@ucs.orst.edu Tue Jun 04 01:09:51 1996
Received: from ucs.orst.edu (root@UCS.ORST.EDU [128.193.4.5]) by tapr.org
(8.7.5/8.7.3/1.9) with SMTP id BAA17355 for <HFSIG@TAPR.ORG>; Tue, 4 Jun 1996
01:09:10 -0500 (CDT)
Received: from p08.t0.monrotel.com by ucs.orst.edu;
(5.65v3.2/1.1.8.2/13Mar96-1233PM)
     id AA13543; Mon, 3 Jun 1996 23:08:10 -0700
Message-Id: <1.5.4.16.19960604073016.317f6a88@ucs.orst.edu>
X-Sender: forrerj@ucs.orst.edu
X-Mailer: Windows Eudora Light Version 1.5.4 (16)
Mime-Version: 1.0
Content-Type: text/plain; charset="us-ascii"
Date: Mon, 03 Jun 1996 23:30:16 -0800
To: HFSIG@tapr.org
From: Johan Forrer <forrerj@ucs.orst.edu>
Subject: Root raised cosine pulse shaping.

Hi Tom,

Well ... After fixing a killer bug, the root raised cosine pulse shaping now works like it is supposed to. The nicest, tightest constellation points that I have seen thus far and no problem with clock or carrier extraction either. Very nice eye patterns - I still have to implement pulse shaping filters in the receiver to gain all the good benefits due to bandwidth reduction. I am getting excellent results with a rolloff factor of 0.6.

Thanks for all the good hints.

--Johan

From BRYD@KIDD.CO.ZA Tue Jun 04 01:12:50 1996
Received: from igw01 (igw01.kidd.co.za [192.96.246.1]) by tapr.org

(8.7.5/8.7.3/1.9) with SMTP id BAA17391 for <hfsig@tapr.org>; Tue, 4 Jun 1996 01:12:47 -0500 (CDT)
Received: from KIDD.CO.ZA by igw01 with smtp
(Smail3.1.29.1 #3) id m0uQpMC-000P9BC; Tue, 4 Jun 96 08:12 GMT+0200
Received: from KenMail-Message_Server by KIDD.CO.ZA
with Novell_GroupWise; Tue, 04 Jun 1996 08:16:00 +0200
Message-Id: <s1b3f0c0.001@KIDD.CO.ZA>
X-Mailer: Novell GroupWise 4.1
Date: Tue, 04 Jun 1996 20:08:03 +0200
From: Danie Brynard <BRYD@KIDD.CO.ZA>
To: hfsig@tapr.org
Subject: [HFSIG:1171] Re: Center freq for rtty ? -Reply

...besides some DSP algorithms that works better when you have more cycles of the waveform within a signaling element.....

I see. Well I will try to implement both and see if there are any practical differences. I noted that Dave W3HCF is also using the high tones in his DSP-93 implementation. Dave any comments from your side ?

73 Danie zs6awk

From karn@qualcomm.com Tue Jun 04 01:33:29 1996
Received: from servo.qualcomm.com (servo.qualcomm.com [129.46.128.14]) by tapr.org (8.7.5/8.7.3/1.9) with ESMTP id BAA18009 for <hfsig@tapr.org>; Tue, 4 Jun 1996 01:33:26 -0500 (CDT)
Received: (from karn@localhost) by servo.qualcomm.com (8.7.5/1.0/8.7.2/1.8) id XAA01122; Mon, 3 Jun 1996 23:32:54 -0700 (PDT)
Date: Mon, 3 Jun 1996 23:32:54 -0700 (PDT)
From: Phil Karn <karn@qualcomm.com>
Message-Id: <199606040632.XAA01122@servo.qualcomm.com>
To: hfsig@tapr.org
In-reply-to: <199606040428.XAA22147@hilly.apci.net> (silbaugh@apci.net)
Subject: Re: [HFSIG:1179] Re: QPSK BANDWIDTH ?

Eric,

An excellent writeup. One nit:

>cosine filetering. Straight QPSK at T bauds/sec should
>have the classic sin(x)/x squared power spectrum with a
>main lobe width (null-to-null) of T Hz. What you

Shouldn't that be 2T Hz from null to null? E.g., a 1200 baud (2400 bps) QPSK signal with carrier at 1600 Hz would have first nulls at 400 Hz and 2800 Hz.

Those damn factors of 2 are so easy to get wrong, especially when some of the textbooks talk about bits per sec and others talk about baud (2 bits/sym for QPSK)...

Phil

From JSANFORD@INFI.NET Tue Jun 04 06:02:05 1996
Received: from mh004.infi.net (mailhost.infi.net [205.219.238.95]) by tapr.org (8.7.5/8.7.3/1.9) with ESMTP id GAA24677 for <hfsig@tapr.org>; Tue, 4 Jun 1996 06:02:03 -0500 (CDT)
Received: from pa1dsp21.orf.infi.net by mh004.infi.net with SMTP
     (Infinet-S-3.3) id HAA13762; Tue, 4 Jun 1996 07:01:28 -0400 (EDT)
Message-Id: <199606041101.HAA13762@mh004.infi.net>
X-Sender: jsanford@infi.net
X-Mailer: Windows Eudora Light Version 1.5.2
Mime-Version: 1.0
Content-Type: text/plain; charset="us-ascii"
Date: Tue, 04 Jun 1996 07:02:15 -0400
To: hfsig@tapr.org
From: Jim Sanford <JSANFORD@INFI.NET>
Subject: Re: [HFSIG:1182] Re: QPSK BANDWIDTH ?

Phil:
Have enjoyed following all the threads you're into . . .

A couple of points:
        1. For what platform are you doing the OQPSK satellite modem? I have 2 dsp-12's and would love to beta test . . .
        2. You made a comment about quadrature outputs from SSB receivers being nice but not available. I believe they will be, SOON. I'm working on a design using Analog Devices DDS's, Mini circuits DBM's, and Maxim's quadrature modulators/demodulators. This thing will initally be for HF, but easily mixed/redesigned for vhf/uhf.
The plummeting cost of the silicon is making this cheaper by the day.

Intend to initally send I and Q to a "stereo" sound card for easy playing with the DSP, but baseband bandwidth on the Maxim demods is large enough that the design will not be limited to audio. Will gladly share details with you, when the prototype is done. Believe cost will be $100 or so for the whole works.

This business is getting terribly exciting -- believe we are on the threshold of some great capabilities at significantly lower prices than Icom, et al, would have us believe. Thanks for all you've done for the state-of-the-art.

73, Jim
wb4gcs@amsat.org

>
>Those damn factors of 2 are so easy to get wrong, especially when some
>of the textbooks talk about bits per sec and others talk about baud (2
>bits/sym for QPSK)...

>
        AMEN!!!!!!!

>Phil
>
>
>

From ssykes@ns2.emirates.net.ae Tue Jun 04 07:36:14 1996
Received: from ns2.emirates.net.ae (ns2.emirates.net.ae [194.170.1.7]) by tapr.org (8.7.5/8.7.3/1.9) with SMTP id HAA27539 for <hfsig@tapr.org>; Tue, 4 Jun 1996 07:36:07 -0500 (CDT)
Received: from csb059.emirates.net.ae by ns2.emirates.net.ae (5.x/SMI-SVR495081401)
     id AA22662; Tue, 4 Jun 1996 16:35:50 +0400
Received: by csb059.emirates.net.ae with Microsoft Mail
     id <01BB5233.827F2140@csb059.emirates.net.ae>; Tue, 4 Jun 1996 16:33:06 +-400
Message-Id: <01BB5233.827F2140@csb059.emirates.net.ae>
From: Stephan Sykes <ssykes@ns2.emirates.net.ae>
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Subject: RE: [HFSIG:1168] Center freq for rtty ?
Date: Tue, 4 Jun 1996 16:25:54 +-400
Encoding: 29 TEXT, 59 UUENCODE
X-Ms-Attachment: WINMAIL.DAT 0 00-00-1980 00:00

I was told that the reason for the high tones was to place the harmonics of the tone outside of the IF filter bandwidth. Improved filters and DSP make this unnecessary now.

Steve Sykes
KD2OM/A61AA

From: Danie Brynard[SMTP:BRYD@kidd.co.za]
Sent: Monday, June 03, 1996 12:30 PM
To: hfsig@tapr.org
Subject: [HFSIG:1168] Center freq for rtty ?

By the way...

Why was such high tones selected for RTTY ? (2125Hz mark, 2295Hz space) My hf rig shows much better response at 800 to 1000Hz. Would it not be better to implement the channel filters for a RTTY DSP modem around say 800Hz ?

I am busy with a rtty dsp modem for the evm and is now wondering about some of my earlier choices :-) From a DSP point of view would it not be better to work around 800Hz ? I have used my PK232 with great success in the past using my 500Hz CW filter on RTTY.

danie zs6awk

```
begin 600 WINMAIL.DAT
M>)\^(@@,`0:0" `$````````!``$``0>0!@`(````Y 0````````#H``$-@ 0`
M`@````(``@`!!) &`$ #````"````% `````,``# %`````"P`/#@``````"`?\/
M`0```#L``````````@2L?I+ZC$!F=;@#=`0]4`@````!H9G-I9T!T87!R+F]R
M9P!33510`&AF<VEG0'1A<'(N;W)G```>``(P`0````4```!33510``````!X`
M`S !`````#P```&AF<VEG0'1A<'(N;W)G```>`!H``0```!0```!215!/4E0N
M25!-+DY/5$4N3D12`$ `,@"@$]'L$%*[`0,`! P```````P`%#/____\#`!4,
M````$ ,`_@\&`````'@`!$ $````!$````3F\@=')A;G-P;W)T('!R;W9I9&5R
M('=A<R!A=F%I;&%B;&4@9F]R(&1E;&EV97)Y('1O('1H:7,@<F5C:7!I96YT
M+@`>``$P`0```!$`````G:&9S:6= =&%P<BYO<F<G```````(!"S !`````% ``
M`%--5% Z2$9324= 5$%04BY/4D<``P``.0``````"`1(Z`0```#L``````````
M@2L?I+ZC$!F=;@#=`0]4`@````!H9G-I9T!T87!R+F]R9P!33510`&AF<VEG
M0'1A<'(N;W)G```>`!,Z`0```!$`````G:&9S:6= =&%P<BYO<F<G```````(!
M%#H!`````% ```%--5% Z2$9324= 5$%04BY/4D<`"P! .@$`````"`?8/`0``
M``0`````````%#@`````,``# &`````"P`/#@$`````"`?\/`0```#L``````````
M@2L?I+ZC$!F=;@#=`0]4`@````!H9G-I9T!T87!R+F]R9P!33510`&AF<VEG
M0'1A<'(N;W)G```>``(P`0````4```!33510``````!X``S !`````#P```&AF
M<VEG0'1A<'(N;W)G```>`!H``0`````@```!)4$TN3D]410,`%0P!``````P#^
M#P8`````>``$P`0```!$`````G:&9S:6= =&%P<BYO<F<G```````(!"S !````
M% ```%--5% Z2$9324= 5$%04BY/4D<``P``.0``````"`10Z`0```! ```!F
M!!_Z2KW/$;"Y1$535 ``"P! .@$`````"`?8/`0`````0`````````&=:(!"( '
M`!@```!)4$TN36EC<F]S;V9T($UA:6PN3F]T90`Q" $$@ $`* ```%)%.B!;
M2$9324<Z,3$V.%T@0V5N=&5R(&9R97$@9F]R(')T='D@/P`L# $%@ ,`#@``
M`,P'!@`$`! `&0`V``(`/@$!!@`'`!@```!)4$TN36EC<F]S;V9T($UA:6PN
M3F]T90`Q" $@@ ,`#@```,P'!@`$``8`$@`)``(```$!"8 !`"$````V-C T
M,49&031!0D1#1#1C$Q0C!".30T-#4U,S4T,# P, `#!P$#D 8`5 4``!,````+
M``",````````,`)@```````"P`I``$````#`#8```````$ `.0! X8WY$%*[`1X`
M< `!````* ```%)%.B!;2$9324<Z,3$V.%T@0V5N=&5R(&9R97$@9F]R(')T
M='D@/P`"`7$``0```!8````!NU&\F=_Z'P1GO4H1S["Y1$535 `````>`!X,
M`0````4```!33510``````!X``'PP!`````%P```'-S>6ME<T!T:6%P<BYE<RYN
M970N864```,`!A DK%B`P``'$*\"```>``@0`0```&4```!)5T%35$%$5B,1(
M05142$5214%T%33TY&3U)(42$5(24=(5$%04BYP4-705-43U!,04)%5%,!%2$%234].
M24-%5E/542$5$9543TE&4$%E%2&2$5&=%8Q)ANE2(0'T4E4$5$5%A(5%$2
M``````(!"1 !````J0,``*4#```X!I@``3%I&=>>RL$<G_``H!#P(5`J@%ZP*#P4@
M`% "\@`&`/S,4&;0X+!P/0*`!6!0M-(3,1'"3!$M+!5-0(YG=A8L`1BRQNY`&$PZ
M`H]6`1%:A`!.K]@EYT7Q8K*ZO>F`%L4M,&4\&`!R+A!XK&@-&O`.O-T2TM<26I\Y`(A
M8T!$0$<&*!*#P*:(!^N"8(<9\$.EX\&!2/G>N\J=J%S0>\ORCQE$+$>?DH?Q`<M9
M+8$/Z%AH5%;+,$W`6\/\N+3U>$B`>@Q!W5K0!R;GZJ&5$7@Q`7S=8<&SB%=>-)%S"2
M2YGY_9&!`!OKV$I\=P.$D\:*6V!,``'5OF@$A0E48:)#,<Y_@`A%5@)[TBUD=^SL1G
M3$T!Y`<FVE$<&>"]BWRV&C\@0"X+=R_&!`_RN`2F%:ENHM/4[=A>7)E`;ME`%P-R
M5Z'E\&-O,&2M&W/90E5D;S9FA)V:!`1@H@1)^_"OCD<`[T@?`E"JM(?T<>HC\"&`"L
M:)M4I`&@(@!_B#`":S9\BLB@4ER=G=D)$J-"@G2`!RTU51Y%7&FW\,DE"CH;P$!U&`(
MM&\Z?Z+<<XE@X8<?.Z0;%QE`\Q%$:'`E^M\2"A'>@`02%PF>7G@$!NUCX$MCZ'
M"+Q;(9$ALG[?'!'QU`\N'I,#<[=O:!%+'>P&G:P+3 $"&R$K%I%&]M:'C?F_C\+N
MD+=!-%4E<MT\'PVJNUD0'_9&!!3D;6F]]O!O/5@5LOVL>N?4D7D=QBT6:4>94'HL
M-"*!+)QRTM_D8$,#V$RD0/5C)(WR^;;'X&CL;!I!4:\JG\25$/Y/15AH)\-8.3@/RS
M.U:H`+`!V\#4R-)!T]3:/^J&6VQ!$A(>(8L15VG12P&"SB!-,$9(0(%6[:V(0@$Q
M!R`KXV=_A3]'D@B$#!R`$A\O)@O/`JM4^Y!6P6.7QEY>JSRLF4N]OJ9
```

M(:$1<&\?0 >1.BU.,'\O$E7A(^)04 N !4 ?@7;_")!<LE'Q4H$*A5+/7-!,
MT/]75E@5&N$1@";16< B\5ZQ\%!+,C,1X%H#"<$;P/]-=DH`)0)2@ .@&_(*
ML!/ EV5173%>HC58(T-7(58_''%+PB)@)<PWL#%2>G/P-F%W:R7,13]&3RP%
M"PJ%%3$`<' ````#`! 0```````,`$1 `````0 `', #JTQ*\4;L!0 `(, #J
MTQ*\4;L!`@$4.@$````0````9@0?^DJ]SQ&PN41%4U0``!X`/0`!````!0``
+`%)%.B ``````(P`
`
end

From LANIER.R.A-@smtpgty.bwi.wec.com Tue Jun 04 08:44:53 1996
Received: from tron.bwi.wec.com (tron.bwi.wec.com [129.228.4.1]) by tapr.org (8.7.5/8.7.3/1.9) with SMTP id IAA29742 for <hfsig@tapr.org>; Tue, 4 Jun 1996 08:44:41 -0500 (CDT)
Received: from smtpgty.bwi.wec.com by tron.bwi.wec.com; (5.65/1.1.8.2/31May95-0229PM)
id AA19449; Tue, 4 Jun 1996 09:42:44 -0400
Received: from ccMail by smtpgty.bwi.wec.com
(IMA Internet Exchange 2.0 Enterprise) id 1B43D890; Tue, 4 Jun 96 09:43:37 -0400
Mime-Version: 1.0
Date: Tue, 4 Jun 1996 09:43:15 -0400
Message-Id: <1B43D890.1858@smtpgty.bwi.wec.com>
From: LANIER.R.A-@smtpgty.bwi.wec.com (LANIER.R.A-)
Subject: Re: [HFSIG:1156] Re: DSP for Linux (was: Modem audio sample)
To: hfsig@tapr.org
Content-Type: text/plain; charset=US-ASCII
Content-Transfer-Encoding: 7bit
Content-Description: cc:Mail note part

Johan,

I would interested in the URLs you are speaking about. Also, what is the "size" of the box you are refering to?

Tony, KE4ATO

______________________________ Reply Separator _________________________________
Subject: [HFSIG:1156] Re: DSP for Linux (was: Modem audio sample)
Author: hfsig@tapr.org at BALT.SMTP
Date: 5/23/96 9:02 PM

Hi Steve,

Another gem for Linux is Matcom; its a Matlab to C++ translator. Then there is Octave, which is a Matlab look-alike. Of course this software will only mean something to you if you are familiar with Matlab. Matlab allows for pretty sophisticated DSP evaluation and simulation. Also don't forget about Ptolemy - really nice but you need a fairly big Linux box to use it.

I don't have the URL's available at the moment, but can let you know if there is interest.

--Johan

At 01:55 PM 5/23/96 -0500, you wrote:
>At 07:22 AM 5/23/96 -0500, you wrote:
>>   Steve,
>>
>>   Where do you find this great stuff???
>
>The best way to keep up with what's new with Linux is to read the
>comp.os.linux.announce newsgroup. There's also a web site that archives
>them, the URL is http://www.cs.helsinki.fi/%7Ewirzeniu/linux/cola.html.
>
>Otherwise, I don't watch a lot of TV. I'd much rather read and experiment :-).
>
>73,
>
> - Steve, N7HPR
>
>srbible@gnatnet.net
>n7hpr@amsat.org
>n7hpr@tapr.org
>
>

From FORRERJ@frl.orst.edu Tue Jun 04 10:36:46 1996
Received: from cornus.FSL.ORST.EDU (root@FSL.ORST.EDU [128.193.112.105]) by tapr.org (8.7.5/8.7.3/1.9) with SMTP id KAA05172 for <hfsig@tapr.org>; Tue, 4 Jun 1996 10:36:44 -0500 (CDT)
Received: from frl.orst.edu (FRL.ORST.EDU [128.193.118.10]) by cornus.FSL.ORST.EDU (8.6.9/8.6.9) with ESMTP id IAA29269 for <hfsig@tapr.org>; Tue, 4 Jun 1996 08:36:41 -0700
Received: from FRL/SpoolDir by frl.orst.edu (Mercury 1.21);
    4 Jun 96 08:36:46 PST8PDT
Received: from SpoolDir by FRL (Mercury 1.21); 4 Jun 96 08:36:35 PST8PDT
From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Organization: Forest Research Lab. Oregon State
To: hfsig@tapr.org
Date: Tue, 4 Jun 1996 08:36:30 -0800
Subject: Re: [HFSIG:1177] Re: QPSK BANDWIDTH ?
Priority: normal
X-mailer: Pegasus Mail v3.22
Message-ID: <59E4CAB540F@frl.orst.edu>

Tom, Phil and Eric,

Much thanks for the excellent ideas and feedback.

I do suspect that some of the spectral lines I'm seeing with my QPSK signal is due to the nature of the cyclic patters that is being sent. I'll work on a PRBS generator as suggested.

Clock and carrier extraction is a fascinating subject and I'll certainly

look at Phil's ideas. For this HF modem, the symbol rate is so low that
FFT's and all sort of things alike may be put to good use, and
I'll get to that shortly. The main thing though is that HF is not too
friendly to true synchronous detection methods - the key to success is to go
for robustness in algorithm design. I think I have some ideas based on
robustness (in the statistical sense) that seem work reasonably well,
but need to be put through the wringer.

I have been looking at epoch synch methods but was seeing enough drift even
with two similar sound cards over a few seconds of signal. However, over a
frame duration of a second or two, I suspect epoch sync may just work out
fine.

Keep up the good work.

--Johan

From forrerj@ucs.orst.edu Tue Jun 04 10:50:48 1996
Received: from ucs.orst.edu (forrerj@UCS.ORST.EDU [128.193.4.5]) by tapr.org
(8.7.5/8.7.3/1.9) with SMTP id KAA05759 for <hfsig@tapr.org>; Tue, 4 Jun 1996
10:50:44 -0500 (CDT)
Received: by ucs.orst.edu; (5.65v3.2/1.1.8.2/13Mar96-1233PM)
id AA02943; Tue, 4 Jun 1996 08:50:37 -0700
Date: Tue, 4 Jun 1996 08:50:37 -0700 (PDT)
From: Johan Forrer <forrerj@ucs.orst.edu>
To: hfsig@tapr.org
Subject: Re: [HFSIG:1183] Re: QPSK BANDWIDTH ?
In-Reply-To: <199606041101.HAA13762@mh004.infi.net>
Message-Id: <Pine.OSF.3.91.960604084018.15342A-100000@ucs.orst.edu>
Mime-Version: 1.0
Content-Type: TEXT/PLAIN; charset=US-ASCII

Hi Jim,

Sounds interesting. Would you care to elaborate more?

I am sort of midway in a project building a VHF/UHF digital receiver for
satellite work. It uses a CATV tuner for the RF front-end which has a 45MHz
IF, 20 MHz A/D fed into a Harris DDC that feeds I/Q outputs into a stereo
CODEC connected to a EZ-kit (ADSP-2181). I use an AD7008 DSS and a PLL for
frequency control.

I thought I would stay away from HF for a while until I learn a bit more
about the RF digital dynamic range, AGC, and it's affects. Guess there is a
lot to learn.

On Tue, 4 Jun 1996, Jim Sanford wrote:

> Phil:
> Have enjoyed following all the threads you're into . . .
>
> A couple of points:
>   1. For what platform are you doing the OQPSK
> satellite modem? I have 2 dsp-12's and would love to
> beta test . . .
>   2. You made a comment about quadrature outputs
> from SSB receivers being nice but not available. I believe
> they will be, SOON. I'm working on a design using
> Analog Devices DDS's, Mini circuits DBM's, and Maxim's
> quadrature modulators/demodulators. This thing will initally
> be for HF, but easily mixed/redesigned for vhf/uhf.
> The plummeting cost of the silicon is making this cheaper
> by the day.

I would be interested to see what sideband rejection youre going to get.
Don't be dissapointed if it's only about 40 dB.

> > Intend to initally send I and Q to a "stereo" sound card for
> easy playing with the DSP, but baseband bandwidth on the
> Maxim demods is large enough that the design will not be
> limited to audio. Will gladly share details with you, when
> the prototype is done. Believe cost will be $100 or so for
> the whole works.
>
> This business is getting terribly exciting -- believe we are on
> the threshold of some great capabilities at significantly lower
> prices than Icom, et al, would have us believe. Thanks for al

--Johan, KC7WW

From dibene@VNET.IBM.COM Tue Jun 04 10:58:13 1996
Received: from VNET.IBM.COM (vnet.ibm.com [199.171.26.4]) by tapr.org
(8.7.5/8.7.3/1.9) with SMTP id KAA05916 for <hfsig@tapr.org>; Tue, 4 Jun 1996
10:58:07 -0500 (CDT)
Message-Id: <199606041558.KAA05916@tapr.org>
Received: from ROMVMNIC by VNET.IBM.COM (IBM VM SMTP V2R3) with BSMTP id 8939;
   Tue, 04 Jun 96 11:57:49 EDT
Date: Tue, 4 Jun 96 17:55:39 EDT
From: "Alberto di Bene (xx39-2-596.25744)" <dibene@VNET.IBM.COM>
To: hfsig@tapr.org
Subject: Tom's books

> For those that might be wondering what this is about; I have been
> exploring some of the materials in Tom's soon-to-be-released book on
> modem design. The section on pulse shaping is particularly good and

 Now you got me curious... any chance to have the title and ISBN of that
 book ? Thanks.

 73 de

Alberto di Bene, I2PHD

From cbuttsch@slonet.org Tue Jun 04 12:37:32 1996
Received: from spork.callamer.com (cbuttsch@spork.callamer.com [199.74.141.2]) by tapr.org (8.7.5/8.7.3/1.9) with ESMTP id MAA09926 for <hfsig@tapr.org>; Tue, 4 Jun 1996 12:36:56 -0500 (CDT)
Received: from localhost (cbuttsch@localhost) by spork.callamer.com (8.7.5/8.7.3) with SMTP id KAA22180; Tue, 4 Jun 1996 10:33:44 -0700 (PDT)
Date: Tue, 4 Jun 1996 10:33:42 -0700 (PDT)
From: Clifford Buttschardt <cbuttsch@slonet.org>
X-Sender: cbuttsch@spork.callamer.com
To: Stephan Sykes <ssykes@ns2.emirates.net.ae>
cc: Bill DeCarle <bill@ietc.ca>, hfsig@tapr.org
Subject: Re: [HFSIG:1184] RE: Center freq for rtty
In-Reply-To: <01BB5233.827F2140@csb059.emirates.net.ae>
Message-ID: <Pine.SOL.3.93.960604102455.17919D-100000@spork.callamer.com>
MIME-Version: 1.0
Content-Type: TEXT/PLAIN; charset=US-ASCII

HI All. I was one of probably many that indicated that the high tones were used to keep harmonics down when using AFSK in SSB rig. Indeed, nowdays there is no reason to use such high tones and commercially, those higher tones are obsolete! We used such high frequencies as we simply duplicated telephone channel filters in which 3 KHz was the upper limit. Not a one of us knew enough about fileter design to change the system!

I fully concer with the use of 800 Hz as a center frequency. All of Bill DeCarles work is centered there as is all my filter designs. It seems a good selection although a short time ago I was thumping for 500 Hertz as it would have make synthesizer design simpler...I lost!!

Danie- I have no way of knowing what you said in the Winmail portion of your note. Can't cope with that! sorry Cliff Buttschardt W6HDO/K7RR

On Tue, 4 Jun 1996, Stephan Sykes wrote:

> I was told that the reason for the high tones was to place the harmonics of
> the tone outside of the IF filter bandwidth. Improved filters and DSP make
> this unnecessary now.
>
> Steve Sykes
> KD2OM/A61AA
>
> ----------
> From: Danie Brynard[SMTP:BRYD@kidd.co.za]
> Sent: Monday, June 03, 1996 12:30 PM
> To: hfsig@tapr.org
> Subject: [HFSIG:1168] Center freq for rtty ?
>
> By the way...
>
> Why was such high tones selected for RTTY ? (2125Hz mark, 2295Hz
> space) My hf rig shows much better response at 800 to 1000Hz. Would
> it not be better to implement the channel filters for a RTTY DSP modem
> around say 800Hz ?

>
> I am busy with a rtty dsp modem for the evm and is now wondering
> about some of my earlier choices :-) From a DSP point of view would it
> not be better to work around 800Hz ? I have used my PK232 with great
> success in the past using my 500Hz CW filter on RTTY.
>
> danie zs6awk
>
>
>
>
> begin 600 WINMAIL.DAT

```
> M>)\^(@@,`0:0" `$````````!``$``0>0!@`(````Y 0````````#H``$-@ 0`
> M`@````(``@`!!) &`$ #````"`````% `````,``# %`````"P`/#@``````"`?\/
> M`0```#L`````````@2L?I+ZC$!F=;@#=`0]4`@`````!H9G-I9T!T87!R+F]R
> M9P!33510`&AF<VEG0'1A<'(N;W)G```>``(P`0````4```!33510``````!X`
> M`S !`````#P```&AF<VEG0'1A<'(N;W)G```>`!H``0```!0```!215!/4E0N
> M25!-+DY/5$4N3D12`$ `,@"@$]'L$%*[`0,`! P```````P`%#/____\#`!4,
> M`````$ ,`_@\&`````'@`!$ $````!$`````3F\@=')A;G-P;W)T('!R;W9I9&5R
> M('=A<R!A=F%I;&%B;&4@9F]R(&1E;&EV97)Y('1O('1H:7,@<F5C:7!I96YT
> M+@`>``$P`0```!$`````G:&9S:6= =&%P<BYO<F<G``````(!"S !`````% ``
> M`%--5% Z2$9324= 5$%04BY/4D<``P``.0``````"`1(Z`0```#L``````````
> M@2L?I+ZC$!F=;@#=`0]4`@`````!H9G-I9T!T87!R+F]R9P!33510`&AF<VEG
> M0'1A<'(N;W)G```>`!,Z`0```!$````G:&9S:6= =&%P<BYO<F<G``````(!
> M%#H!`````% ```%--5% Z2$9324= 5$%04BY/4D<`"P! .@$`````"`?8/`0``
> M``0``````````%#@`````,``# &`````"P`/#@$`````"`?\/`0```#L``````````
> M@2L?I+ZC$!F=;@#=`0]4`@`````!H9G-I9T!T87!R+F]R9P!33510`&AF<VEG
> M0'1A<'(N;W)G```>``(P`0````4```!33510``````!X``S !`````#P```&AF
> M<VEG0'1A<'(N;W)G```>`!H``0`````@```!)4$TN3D]410,`%0P!``````P#^
> M#P8````>``$P`0```!$`````G:&9S:6= =&%P<BYO<F<G``````(!"S !`````
> M% ```%--5% Z2$9324= 5$%04BY/4D<``P``.0``````"`10Z`0```! ```!F
> M!!_Z2KW/$;"Y1$535 ``"P! .@$`````"`?8/`0````0``````````&=:(!"( '
> M`!@```!)4$TN36EC<F]S;V9T($UA:6PN3F]T90`Q" $$@ $`* ```%)%.B!;
> M2$9324<Z,3$V.%T@0V5N=&5R&5R(&9R97$@9F]R(')T=''D@/P`L# $%@ ,`#@``
> M`,P'!@`$`! `&0`V``(`/@$!!!@`'`!@```!)4$TN36EC<F]S;V9T($UA:6PN
> M3F]T90`Q" $@@ ,`#@```,P'!@`$``8`$@`)``(```$!"8 !`"$8 !`"$````V-C T
> M,49&031!0D1#1C$Q0C!"".30T-#4U,S4T,# P, `#!P$#D 8`5 4``!,````+
> M`",````````,`)@``````"P`I``$````#`#8``````$ `.0! X8WY$%*[`1X`
> M< `!````* ```%)%.B!;2$9324<Z,3$V.%T@0V5N=&5R&5R(&9R97$@9F]R(')T
> M='D@/P`"`7$``0``!8````!NU&\F=_Z'P1GO4H1S["Y1$535 `````>`!X,
> M`0````4```!33510`````!X`'PP!````%P```'-S>6ME<T!E;6ER871E<RYN
> M970N864```,`!A DJK%B`P`'$*\"```>``@0`0```&4```!)5T%35$],1%1%
> M05142$5214%33TY&3U)43$52$5(24=(5$].5$0U-%$A%U,$8%$A%2$%234].1%,234].
> M24-33T942$%543TY%544TE$$14]&5$4A%249&24Q415)/04Y$)"04Y$5TE$5$$A)35!2
> M``````(!"1 !````J0,``*4#``X!@``3%I&=>>RL$<G_``H!#P(5`J@%ZP*#
> M`% "\@D"`&-H"L!S970R-P8`!L,"@S(#Q@<3`H,S$6\"@`T1B+0B@R"$C@
> M`H!]"H`(SPG9.U_$6#S(U-0*`#"/H!#+A"P@4(&BC-\"@ ,`$0<3$K=)@/T%@@/
> M;&$(T0H@.',!:2 #!T!%P#J!*H+@";X9IR1#1P2S!A7)A0A$P'HB@&"5"&P@EU8I`@K)0+@`-%&!
> M( M18QP0'.,*P 1@Y0,`8P8Z0/8'"`/P",V(&5E8P;V@8J@<71W "<1&D\0-&!
> MP&&`(!0<&1=Q#D&L@4%`$4-K&(5P` P `.2Q1`@M1DDN1F*H8( @LE#@@)&H0.L)P
> M%V@V*@`$ &$`!& `!'-6YE#%F"J@-6)% 4$0*E<DLA$`6!-@4X/`$`.4L0!3`!O`(
> M0!4@> 3Q+*&N2`=S)6%\2(!4A>YM@8,`A`S!6, GIL2.U`:2, 5>@%'&@6:E4`
> M4S R$;:X8P21)!"RSM7V2!<1QTWWDB*WY4!`&%C@I!-0 GT4I%&`*F0%(%)C!
> M!2Z$PRU@Q06A!5L`B12!@`@#Y#0&%0`$ (T!`!1A!+#X%22`!"`-SR0>+ "4(
> M!0O&`0"0U8.*U&J=K-!(0!`@0`@2%((!@0&,W$`%0`&6'X*8!8H5'P!*@`5T(V
> M*0"!0!,!>5 `\0!BT .0 +('V-D@0!1DAUE
```

```
> M9,(N!: N>F%=+1\N+2\&8 (P+U\P:TT"(&1A\'DL($HDP!P0&4 WX! Q.3DV
> M.( R.C,P,"!033-O+BU4;]LUKS!K: /0'3! `9 34!HN!;!G.6\T?G5B:@<L
> M,3N/,&M;2$9320A'.C$KP#A=($-G-7$AH0-097$<DP`@=,TE8#\H?RF#,S8J
> M]QI%G2OV0B5@&_(;$'DN2&"9)<Q7:"5@&Q)S=1%P.QT:$;!L+#$B\@6Q4E1$
> M5%E$H" H,CC@-21(>B01<FLWX#(RMCE,<0J%<PJP'G I(G"^325@/6 <(!TP
> M2>!H)9#U!"!M2@)B$< ADA803>#_`B 1L".0!4 IX1X"&3 I\+M,@")@5PA@
> M&W *A6D%0/\E@ 5 3[!/IAX1!W +4!/@?S5Q&^,1<231`R C)ARB8?=+M"/C
> M!'%M"H4*P A@([$_)3 E8"GA3(%$O1KP86WY(<!U<TF14I =4%7P1&/^9$W@
> M5I0<ER; 69 CHB21_R6!&P`WD09Q&1!6]@;@(%"W2> #<""C;25@'$!R*;";
> M(:$1<&\?0 >1.BU.,'\O$E7A(^)04 N !4 ?@7;_")!<LE'Q4H$*A5+/7-!,
> MT/]75E@5&N$1@";16< B\5ZQ\%!+,C,1X%H#"<$;P/]-=DH`)0)2@ .@&_(*
> ML!/ EV5173%>HC58(T-7(58_''%+PB)@)<PWL#%2>G/P-F%W:R7,13]&3RP%
> M"PJ%%3$`<' ````#`! 0``````,`$1 `````0 `', #JTQ*\4;L!0 `(, #J
> MTQ*\4;L!`@$4.@$````0````9@0?^DJ]SQ&PN41%4U0``!X`/0`!````!0``
> +`%)%.B ``````(P`
> `
> end
>
>
```


From karn@qualcomm.com Tue Jun 04 15:07:53 1996
Received: from servo.qualcomm.com (servo.qualcomm.com [129.46.128.14]) by tapr.org (8.7.5/8.7.3/1.9) with ESMTP id PAA15704 for <hfsig@tapr.org>; Tue, 4 Jun 1996 15:07:50 -0500 (CDT)
Received: (from karn@localhost) by servo.qualcomm.com (8.7.5/1.0/8.7.2/1.8) id NAA10211; Tue, 4 Jun 1996 13:07:12 -0700 (PDT)
Date: Tue, 4 Jun 1996 13:07:12 -0700 (PDT)
From: Phil Karn <karn@qualcomm.com>
Message-Id: <199606042007.NAA10211@servo.qualcomm.com>
To: hfsig@tapr.org
In-reply-to: <199606041101.HAA13762@mh004.infi.net> (message from Jim Sanford on Tue, 4 Jun 1996 06:10:36 -0500 (CDT))
Subject: Re: [HFSIG:1183] Re: QPSK BANDWIDTH ?

>   1. For what platform are you doing the OQPSK
>satellite modem? I have 2 dsp-12's and would love to
>beta test . . .

The platform is a generic 486 or Pentium PC running my KA9Q NOS TCP/IP package. The modem will be integrated in as an interface driver for a SoundBlaster 16 card *without* any special DSP chips.

Your work with the 'digital friendly' receiver sounds very interesting! I do hope that whatever you come up with can be easily replicated by the average amateur -- in my experience, that's where many promising amateur hardware projects fall down. It's the main reason I've become so interested in using mass-market PC hardware to the greatest possible extent, doing everything else in software. But there are definite limits to the available hardware (particularly in the radios) so your work is most welcome. I agree, this stuff ought not to be as expensive and complicated as it seems to be.

At Dayton I looked at some of the companies selling after-market

crystal filters for commercial SSB transceivers with an eye toward
modifying them for wider baseband bandwidths. Ideally I'd like a ~20
KHz bandwidth so I could fully utilize the A/D bandwidth of a PC sound
card. I could even create a new "IF" at, say, 10 KHz and do my final
"product detection" in DSP software.

The guy behind the Fox-Tango table, being used to selling very sharp
and narrow crystal filters for HF contest and DX use, couldn't figure
out why in the world I would want to make my rig go *wider*. Ah, the
"narrowband mindset" seems pretty well entrenched. :-)

Phil

From LAITINEN@ESHOP.UOREGON.EDU Tue Jun 04 15:49:09 1996
Received: from ESHOP.UOREGON.EDU (eshop.uoregon.edu [128.223.94.14]) by tapr.org
(8.7.5/8.7.3/1.9) with SMTP id PAA17008 for <hfsig@tapr.org>; Tue, 4 Jun 1996
15:49:05 -0500 (CDT)
Date: Sun, 4 Jun 1995 13:48:40 -0700 (PDT)
From: LAITINEN@ESHOP.UOREGON.EDU
Message-Id: <950604134841.b9b@ESHOP.UOREGON.EDU>
Subject: RE: [HFSIG:1189] RE: Center freq for rtty
To: hfsig@tapr.org
X-Vmsmail-To: SMTP%"hfsig@tapr.org"

Gee, I thought that the use of 2125/2975 Hz for AFSK tones in the old
days was related to running AFSK RATT on 2-meters "AM" (and then
eventually FM). Tuning the AFSK generator and TU filters to the proper
frequencies was done using standard tuning forks and lisajous (spelling?)
patterns on an oscilloscope. Standard landline VFCT (Voice Freq Carrier
Telegraph) systems (as I recall) were set up with 85-Hz channels and the
tools (i.e., tuning forks) were around for tuning such things... Thus the
harmonic relationships of the old 2125/2975 tones to these tuning fork
standards were important...

My first 2-meter RATT setup used an ARC-5/T-23 surplus aircraft radio
modulated by an Eico 730K modulator. The RATT gear was a Model 19 with a
W2JAV TU (the one with four neon bulbs on the front).... The receiver was
the 417A converter that was popular in the early 1960's for Project OSCAR...

Larry, WA6JYJ/7

From wd5ivd@tapr.org Tue Jun 04 20:49:23 1996
Received: (from wd5ivd@localhost) by tapr.org (8.7.5/8.7.3/1.9) id UAA29453 for
hfsig@tapr.org; Tue, 4 Jun 1996 20:49:22 -0500 (CDT)
From: Greg Jones <wd5ivd@tapr.org>
Message-Id: <199606050149.UAA29453@tapr.org>
Subject: Re: [HFSIG:1188] Tom's books
To: hfsig@tapr.org
Date: Tue, 4 Jun 1996 20:49:22 -0500 (CDT)
In-Reply-To: <199606041558.KAA05916@tapr.org> from "Alberto di Bene" at Jun 4, 96
11:10:08 am
X-Mailer: ELM [version 2.4 PL25]

Content-Type: text

Title: Wireless Digital Communications: Design and Theory
By: Tom McDermott, N5EG
ISBN: 0-9644707-2-1

Anticipated price will be $39.00

With luck it will be available from TAPR the end of August/first of Sept.
Tom is currently doing small technical corrections and the proof-reader is
working over grammer issues. I get those elemtns and then with luck we go
to thew e printers.

I think there is a web page already on the tapr web server under
publications. If it isn't there -- then I have it ready, but haven't put it
up yet,. Has a complete index...that is if it is up.

This has been a year and a half project. Will be glad to get it finsihed.

Cheers - Greg, WD5IVD

```
>
> >  For those that might be wondering what this is about; I have been
> >  exploring some of the materials in Tom's soon-to-be-released book on
> >  modem design. The section on pulse shaping is particularly good and
>
>  Now you got me curious... any chance to have the title and ISBN of that
>  book ? Thanks.
>
>  73 de
>  Alberto di Bene,  I2PHD
>
>
```

From BRYD@KIDD.CO.ZA Wed Jun 05 01:13:35 1996
Received: from igw01 (igw01.kidd.co.za [192.96.246.1]) by tapr.org
(8.7.5/8.7.3/1.9) with SMTP id BAA14915 for <hfsig@tapr.org>; Wed, 5 Jun 1996
01:13:27 -0500 (CDT)
Received: from KIDD.CO.ZA by igw01 with smtp
     (Smail3.1.29.1 #3) id m0uRBbI-000PEZC; Wed, 5 Jun 96 07:57 GMT+0200
Received: from KenMail-Message_Server by KIDD.CO.ZA
     with Novell_GroupWise; Wed, 05 Jun 1996 08:01:08 +0200
Message-Id: <s1b53ec4.049@KIDD.CO.ZA>
X-Mailer: Novell GroupWise 4.1
Date: Wed, 05 Jun 1996 19:56:05 +0200
From: Danie Brynard <BRYD@KIDD.CO.ZA>
To: hfsig@tapr.org
Subject: [HFSIG:1191] RE: Center freq for rtty -Reply

Larry now that is interesting. Thanks for the bit of history.

danie

From LANIER.R.A-@smtpgty.bwi.wec.com Wed Jun 05 10:56:50 1996
Received: from tron.bwi.wec.com (tron.bwi.wec.com [129.228.4.1]) by tapr.org
(8.7.5/8.7.3/1.9) with SMTP id KAA02516 for <hfsig@tapr.org>; Wed, 5 Jun 1996
10:56:48 -0500 (CDT)
Received: from smtpgty.bwi.wec.com by tron.bwi.wec.com;
(5.65/1.1.8.2/31May95-0229PM)
id AA04947; Wed, 5 Jun 1996 11:53:42 -0400
Received: from ccMail by smtpgty.bwi.wec.com
(IMA Internet Exchange 2.0 Enterprise) id 1B5ADB10; Wed, 5 Jun 96 11:54:25 -0400
Mime-Version: 1.0
Date: Wed, 5 Jun 1996 09:48:54 -0400
Message-Id: <1B5ADB10.1858@smtpgty.bwi.wec.com>
From: LANIER.R.A-@smtpgty.bwi.wec.com (LANIER.R.A-)
Subject: Sine wave PROM
To: hfsig@tapr.org
Content-Type: text/plain; charset=US-ASCII
Content-Transfer-Encoding: 7bit
Content-Description: cc:Mail note part

I am trying to locate information on generating sine wave data for a
EEPROM. I had an article (Radio Electronics) describing this by using
a C program to generate the data. However, I can't find the article.

Can someone help with this problem?

73s de
Tony, KE4ATO

From silbaugh@apci.net Wed Jun 05 22:39:35 1996
Received: from hilly.apci.net (root@hilly.apci.net [206.100.36.3]) by tapr.org
(8.7.5/8.7.3/1.9) with SMTP id WAA26548 for <hfsig@tapr.org>; Wed, 5 Jun 1996
22:39:32 -0500 (CDT)
Received: from dialup167.apci.net (dialup167.apci.net [206.100.36.167]) by
hilly.apci.net (8.6.12/8.6.9) with SMTP id WAA07740 for <hfsig@tapr.org>; Wed, 5
Jun 1996 22:40:02 -0500
Date: Wed, 5 Jun 1996 22:40:02 -0500
Message-Id: <199606060340.WAA07740@hilly.apci.net>
X-Sender: silbaugh@apci.net
X-Mailer: Windows Eudora Light Version 1.5.2
Mime-Version: 1.0
Content-Type: text/plain; charset="us-ascii"
To: hfsig@tapr.org
From: "Eric E. Silbaugh" <silbaugh@apci.net>
Subject: Re: QPSK BANDWIDTH ?

Phil,

Good nit! Fourier is probably turning in his grave.

I was visualizing the spectra at baseband, wrote about it as
RF modulated, and forgot to multiply the bandwidths by two.
Arrrgggh! Thus, the first null in a QPSK signal of T baud will

be T Hz away from the carrier; the main lobe width is 2T Hz;
and the bandwidth with raised cosine filtering, rolloff factor
of 0.2, will be 2.4T Hz.

Did I get it right this time?

Eric, N2NNP

From FORRERJ@frl.orst.edu Thu Jun 06 13:20:28 1996
Received: from cornus.FSL.ORST.EDU (root@FSL.ORST.EDU [128.193.112.105]) by tapr.org (8.7.5/8.7.3/1.9) with SMTP id NAA29406 for <hfsig@tapr.org>; Thu, 6 Jun 1996 13:20:23 -0500 (CDT)
Received: from frl.orst.edu (FRL.ORST.EDU [128.193.118.10]) by cornus.FSL.ORST.EDU (8.6.9/8.6.9) with ESMTP id LAA10445 for <hfsig@tapr.org>; Thu, 6 Jun 1996 11:20:19 -0700
Received: from FRL/SpoolDir by frl.orst.edu (Mercury 1.21);
    6 Jun 96 11:20:22 PST8PDT
Received: from SpoolDir by FRL (Mercury 1.21); 6 Jun 96 11:20:03 PST8PDT
From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Organization: Forest Research Lab. Oregon State
To: hfsig@tapr.org
Date: Thu, 6 Jun 1996 11:20:01 -0800
Subject: Further thoughts on pulse shaping for HF
Priority: normal
X-mailer: Pegasus Mail v3.22
Message-ID: <5D1078A41EC@frl.orst.edu>

Hi folks,

Another thought/question on pulse shaping for HF;

Consider the recovered signals from a QPSK modem where the final product is a raised cosine - I do encounter a little annoying sideffect. In the vacinity of the sampling point, one has to read both I and Q - for QPSK case, one expects that either I or Q must be near zero, i.e. a pair from the set {1,0 -1,0 0,1 0,-1 for I/Q}.

Unfortunately, however, the pulse shaping procedure forces part of the pulse beyond the +/-T mark, to go negative, which, combined with a bit of jitter introduced by the channel effects, causes the channel that is supposed to be zero, to an off-zero value. These uncertainties are further aggravated that this error in the zero value, changes fairly rapidly in the vacinity of the sampling point. It would have been nice if things around the zero transition were a bit more stable.

Now, if I were to use another shape function like Dolph-Chebychef, or the Blackman-Harris shape for minimum sidelobes, instead of raised cosines, I would not have true nyquist pulses - the tradeoff is a some of ISI, however, would'nt the result be more robust against this jitter problem as described above. I.e., would'nt the tradeoff between ML detection and robustness for clock/carrier extraction, which is more crucial, make sense?

Comments or suggestions would be appreciated.

--Johan, KC7WW

From karn@qualcomm.com Thu Jun 06 20:19:35 1996
Received: from servo.qualcomm.com (servo.qualcomm.com [129.46.128.14]) by tapr.org (8.7.5/8.7.3/1.9) with ESMTP id UAA14417 for <hfsig@tapr.org>; Thu, 6 Jun 1996 20:19:32 -0500 (CDT)
Received: (from karn@localhost) by servo.qualcomm.com (8.7.5/1.0/8.7.2/1.9) id SAA15345; Thu, 6 Jun 1996 18:18:57 -0700 (PDT)
Date: Thu, 6 Jun 1996 18:18:57 -0700 (PDT)
From: Phil Karn <karn@qualcomm.com>
Message-Id: <199606070118.SAA15345@servo.qualcomm.com>
To: hfsig@tapr.org
In-reply-to: <199606060340.WAA07740@hilly.apci.net> (silbaugh@apci.net)
Subject: Re: [HFSIG:1195] Re: QPSK BANDWIDTH ?

>I was visualizing the spectra at baseband, wrote about it as
>RF modulated, and forgot to multiply the bandwidths by two.
>Arrrgggh! Thus, the first null in a QPSK signal of T baud will
>be T Hz away from the carrier; the main lobe width is 2T Hz;
>and the bandwidth with raised cosine filtering, rolloff factor
>of 0.2, will be 2.4T Hz.

>Did I get it right this time?

Almost. The null-to-null bandwidth of 2T Hz looks right, but with raised cosine filtering the bandwidth will be *less* than the unfiltered null-to-null bandwidth.

Phil

From karn@qualcomm.com Thu Jun 06 20:32:11 1996
Received: from servo.qualcomm.com (servo.qualcomm.com [129.46.128.14]) by tapr.org (8.7.5/8.7.3/1.9) with ESMTP id UAA14730 for <hfsig@tapr.org>; Thu, 6 Jun 1996 20:32:09 -0500 (CDT)
Received: (from karn@localhost) by servo.qualcomm.com (8.7.5/1.0/8.7.2/1.9) id SAA15427; Thu, 6 Jun 1996 18:31:38 -0700 (PDT)
Date: Thu, 6 Jun 1996 18:31:38 -0700 (PDT)
From: Phil Karn <karn@qualcomm.com>
Message-Id: <199606070131.SAA15427@servo.qualcomm.com>
To: hfsig@tapr.org
In-reply-to: <5D1078A41EC@frl.orst.edu> (FORRERJ@frl.orst.edu)
Subject: Re: [HFSIG:1196] Further thoughts on pulse shaping for HF

>Consider the recovered signals from a QPSK modem where the final product
>is a raised cosine - I do encounter a little annoying sideffect. In the
>vacinity of the sampling point, one has to read both I and Q - for QPSK
>case, one expects that either I or Q must be near zero, i.e. a pair from
>the set {1,0 -1,0 0,1 0,-1 for I/Q}.

Well, you can use any convention you like, but the usual one is to signal with the set { (a,a), (-a,a), (-a,-a), (a,-a) } with a=sqrt(2) times the signal magnitude and with the detectors looking independently at the I and Q components. That is, since I and Q are (ideally) orthogonal the I detector simply produces 1 or -1 (or a soft decision sample somewhere in this range) without caring what is going on in the Q channel. And vice versa.

This particular constellation is used by the 4-phase costas loop, which will drive the I and Q components to equal amplitude. You can of course use the constellation you described, but then you have to rotate your decision regions so they're no longer aligned with the I and Q axes.

Phil

From n4hy@ccr-p.ida.org Fri Jun 07 09:58:16 1996
Received: from idacrd.ccr-p.ida.org (idacrd.ccr-p.ida.org [198.3.41.2]) by tapr.org (8.7.5/8.7.3/1.9) with ESMTP id JAA16703 for <hfsig@tapr.org>; Fri, 7 Jun 1996 09:58:09 -0500 (CDT)
Received: from idacrd.ccr-p.ida.org (daemon@localhost) by idacrd.ccr-p.ida.org (8.7.2/8.7.2) with ESMTP id KAA03102 for <hfsig@tapr.org>; Fri, 7 Jun 1996 10:58:23 -0400 (EDT)
Received: from ccr-p.ida.org (xida.ccr-p.ida.org [198.3.6.62]) by idacrd.ccr-p.ida.org (8.7.2/8.7.2) with SMTP id KAA03098 for <hfsig@tapr.org>; Fri, 7 Jun 1996 10:58:22 -0400 (EDT)
Received: from growler.ccr-p.ida.org (growler.ccr-p.ida.org [198.3.6.3]) by ccr-p.ida.org (8.6.12/8.6.12) with ESMTP id KAA01840 for <hfsig@tapr.org>; Fri, 7 Jun 1996 10:57:41 -0400
From: Bob McGwier <n4hy@ccr-p.ida.org>
Received: (from n4hy@localhost) by growler.ccr-p.ida.org (8.7.5/8.7.3) id KAA00644 for hfsig@tapr.org; Fri, 7 Jun 1996 10:57:40 -0400 (EDT)
Date: Fri, 7 Jun 1996 10:57:40 -0400 (EDT)
Message-Id: <199606071457.KAA00644@growler.ccr-p.ida.org>
To: hfsig@tapr.org
Subject: Re: [HFSIG:1196] Further thoughts on pulse shaping for HF
In-reply-to: <5D1078A41EC@frl.orst.edu> (FORRERJ@frl.orst.edu)

A good approximation to the ML detector for clock AND data together is what you need Johan. Take the final pulse shape (sampled) and its derivative (Guess at this or approximate it. If it is raised cosine, you may compute it).   Call these signals   P(i) and P'(i) , i = -n, -(n-1), ..., 0,1,2,... (n-1),n respectively. Suppose for this exercise that you have an approximate time for the CENTER (the sampling time) for the I and Q channel bauds.

At the putative sampling time t, compute   P*S(t) and P'*S(t). That is, convolve the pulse shape, and its derive with the baseband signal S and do it for S from the I and Q channel. The SIGN of the P*S is the guessed at symbol and SIGN(P*S(t))*tanh [P'*S(t)] = ERROR signal to be multiplied by the gain which will determine your "loop bandwidth". If you are running a first order loop for clock, this signal (gain*ERROR) is applied to the phase of your clock and then you march along to the next symbol time. This is MUCH more robust, stable, etc, than using clock edges, or zero crossings, and so long as you have the pulse shape "about right", even ISI, jitter, does not disturb you much. This was proposed a long time ago, before the advent of DSP devices by Umberto Mengali. It was proven by yours truly that this converges under certain conditions which are near to reality to the maximum likelihood estimator.

A diagram of this might be more useful to nonmathematicians ;=):

```
               Soft Decision -> -> -> Decoder
                   |
                   |
                   |            Data
                   |              |
                   |              |
          ->->->  P*S -> -> {Hard Limiter}  ->-> |
          |                                      |
          |                                      |
S(t)  ->  |                                    Product -> -> ERROR to clock
          |
          |                                      |
          |                                      |
          ->->-> P'*S -> -> -> tanh -> -> -> ->->|
```

The obvious way of combining this for each channel would be to average, or in the case of staggered, run the clock at two times the individual symbol rate and alternate the channels feeding the clock. You may run this as a second order phase locked loop if you believe the clock frequency is also unstable. I used this to derive clock for a really noisy signal from Venus (VEGA Balloon probe). STAY AWAY FROM CLOCK EDGES in a really good modem. Implement the tanh with a table look up and/or some approximation that you find useful. The major condition is that the pulse shape in a idealized channel has zero's at the baud time for neighboring bauds. I have found it to be quite robust. I use Passband equalization whenever possible. The two possible outputs for data depends on whether or not there is error correction coding that needs soft decisions. You will pay larger than a 2 dB penalty to use the hard decision tap for your decoder on HF channels should you decide to go that way. There are other forms of this, but this one is one of the easiest to implement.

Bob

Dr. Robert W. McGwier | n4hy@ccr-p.ida.org: ham radio, scouts,
Center for Communications Research | astronomy, golf (o yea, & math!)

Princeton, N.J. 08520 | Cmte member Troop 5700, ACM Pack 53,
(609)-279-6240(v) (609)-924-3061(f)| Frmr Cncl Comm. GWC 362 Sanhican #2 WWW,
(609)-443-8963 (h) | I used to be a Buffalo . . . NE III-120
Explorer Post 995 advisor | proud parent in Brownie Troop 196

From forrerj@ucs.orst.edu Sat Jun 08 13:18:47 1996
Received: from ucs.orst.edu (root@UCS.ORST.EDU [128.193.4.5]) by tapr.org
(8.7.5/8.7.3/1.9) with SMTP id NAA22005 for <hfsig@tapr.org>; Sat, 8 Jun 1996
13:18:39 -0500 (CDT)
Received: from p00.t0.monrotel.com by ucs.orst.edu;
(5.65v3.2/1.1.8.2/13Mar96-1233PM)
     id AA09996; Sat, 8 Jun 1996 11:18:23 -0700
Message-Id: <1.5.4.16.19960608194055.3f8fec56@ucs.orst.edu>
X-Sender: forrerj@ucs.orst.edu
X-Mailer: Windows Eudora Light Version 1.5.4 (16)
Mime-Version: 1.0
Content-Type: text/plain; charset="us-ascii"
Date: Sat, 08 Jun 1996 11:40:55 -0800
To: hfsig@tapr.org
From: Johan Forrer <forrerj@ucs.orst.edu>
Subject: Re: [HFSIG:1199] Re: Further thoughts on pulse shaping for HF

Hi Bob,

Thanks for sharing your ideas - also welcome to HFSIG.

At 10:00 AM 6/7/96 -0500, you wrote:
>
>
>A good approximation to the ML detector for clock AND data together is what
>you need Johan.

I suspect you are right. The question that have been on my mind lately regards how much of the ideas that we commonly use for synchronous demodulation is worth implementing on HF. As an example, I have been playing with a low baudrate QPSK demodulator over the last few weeks where most of the clock and carrier extraction ideas are based on robust non-syncronous measures. I have had some encouraging results, but everything is not quite working either. The ideas are based roughly on the following two guidelines:

1) Do clock extraction first because
        a) The middle of the baud is where the signal has most energy and
there is
           best phase stability (stay away from the transition zones).
        b) Attempt carrier extraction *only* during this stable period.

2) Decision-directed carrier extraction - preferably not based on PLL's or
Costas loops.

On the latter subject; I have played and struggled with various ideas - what

I found was that the extended Costas loop (i.e., with I/Q sgn and summming of cross products) works very nicely indeed with waveforms that is not overly distorted. In fact in Frerking's book (Digital Signal Processing in Communications Systems, Marvin Frerking ISBN 0442016166 p.444) makes you aware of the fact that you *not* use the output of a matched filter but a lowpass filter to steer the Costas loop. I have found this to be valid.

This is one reason why I'm quite interested in certain constellations that is easier to deal with than others.

>Take the final pulse shape (sampled) and its derivative
>(Guess at this or approximate it. If it is raised cosine, you may compute
>it).   Call these signals   P(i) and P'(i) , i = -n, -(n-1), ..., 0,1,2,...
>(n-1),n respectively. Suppose for this exercise that you have an approximate
>time for the CENTER (the sampling time) for the I and Q channel bauds.
>
>At the putative sampling time t, compute   P*S(t) and P'*S(t). That is,
>convolve the pulse shape, and its derive with the baseband signal S and do
>it for S from the I and Q channel. The SIGN of the P*S is the guessed at
>symbol and SIGN(P*S(t))*tanh [P'*S(t)] = ERROR signal to be multiplied
>by the gain which will determine your "loop bandwidth". If you are running
>a first order loop for clock, this signal (gain*ERROR) is applied to the
>phase of your clock and then you march along to the next symbol time. This
>is MUCH more robust, stable, etc, than using clock edges, or zero crossings,
>and so long as you have the pulse shape "about right", even ISI, jitter,
>does not disturb you much. This was proposed a long time ago, before the
>advent of DSP devices by Umberto Mengali. It was proven by yours truly that
>this converges under certain conditions which are near to reality to the
>maximum likelihood estimator.

Ah .. I like this idea and will give it some more thought. Could you be so kind and give us the full references.

```
>
>
>
>A diagram of this might be more useful to nonmathematicians ;=):
>
>               Soft Decision -> -> -> Decoder
>                   |
>                   |
>                   |            Data
>                   |              |
>                   |              |
>         ->->->  P*S -> -> {Hard Limiter}  ->-> |
>          |                                     |
>          |                                     |
>S(t)  ->  |                                   Product -> -> ERROR to clock
>          |
>          |                                     |
>          |                                     |
```

>   ->->-> P'*S -> -> -> tanh -> -> -> ->->|
>
>The obvious way of combining this for each channel would be to average,
>or in the case of staggered, run the clock at two times the individual symbol
>rate and alternate the channels feeding the clock. You may run this as
>a second order phase locked loop if you believe the clock frequency is also
>unstable. I used this to derive clock for a really noisy signal from
>Venus (VEGA Balloon probe). STAY AWAY FROM CLOCK EDGES in a really good
>modem.

I guess we agree that the usual synchronous demodulation ideas needs careful
rethinking.

>Implement the tanh with a table look up and/or some approximation
>that you find useful. The major condition is that the pulse shape in
>a idealized channel has zero's at the baud time for neighboring bauds.

Would it thus be an advantage to use such a shaping function; I do know that
the Dolph-chebycheff and Blackman-Harris shapes, for example, was designed
for that.

>I have found it to be quite robust. I use Passband equalization whenever
>possible.

How is this done? using adaptive FIR structures? does this require some kind
of training sequence?

>The two possible outputs for data depends on whether or not there
>is error correction coding that needs soft decisions. You will pay larger
>than a 2 dB penalty to use the hard decision tap for your decoder on HF
>channels should you decide to go that way. There are other forms of this,
>but this one is one of the easiest to implement.
>
>Bob
>
>Dr. Robert W. McGwier | n4hy@ccr-p.ida.org: ham radio, scouts,
>Center for Communications Research | astronomy, golf (o yea, & math!)
>Princeton, N.J. 08520 | Cmte member Troop 5700, ACM Pack 53,
>(609)-279-6240(v) (609)-924-3061(f)| Frmr Cncl Comm. GWC 362 Sanhican #2 WWW,
>(609)-443-8963 (h) | I used to be a Buffalo . . . NE III-120
>Explorer Post 995 advisor | proud parent in Brownie Troop 196
>
>
>
>
>

Pardon if I seem too inquisitive and ask too many dumb questions.

--Johan, KC7WW

From hardie@duke.usask.ca Sat Jun 08 14:47:08 1996

Received: from duke.usask.ca (duke.usask.ca [128.233.3.13]) by tapr.org
(8.7.5/8.7.3/1.9) with ESMTP id OAA24958 for <hfsig@tapr.org>; Sat, 8 Jun 1996
14:47:05 -0500 (CDT)
Received: from localhost (hardie@localhost) by duke.usask.ca (8.7.3/8.7.3) with
SMTP id NAA09131 for <hfsig@tapr.org>; Sat, 8 Jun 1996 13:46:57 -0600 (CST)
Date: Sat, 8 Jun 1996 13:46:57 -0600 (CST)
From: Pete Hardie <hardie@duke.usask.ca>
To: hfsig@tapr.org
Subject: Stuck at the beginning

In-Reply-To: <1.5.4.16.19960608194055.3f8fec56@ucs.orst.edu>
Message-ID: <Pine.OSF.3.93.960608132158.5705A-100000@duke.usask.ca>
MIME-Version: 1.0
Content-Type: TEXT/PLAIN; charset=US-ASCII

I'm just starting with DSP and thought (after creating some CW FIR filters) that I'd try to implement a 1200 baud packet TNC on the EZ-KIT Lite.
I had assumed that implementing the modulation side of this was going to be easy but I've run into a major hurdle that I can't figure out.

Generating the NRZI data and modulating it as a continuous phase FSK audio signal was easy. I have set up the DSP program so that it reads KISS packets from the RS-232 port and then generates the FSK. However, if I put the audio from the Kit into either my KAM or Tiny-2 neither of them will decode a packet. I tried this with passall on so that even if the CRC were bad they should print something out - but they don't. In both cases their DCD LED does come on while the packet audio is present so they're seeing something. (BTW - I have used a frequency counter to check the frequency and stability of the generated tones and of the data bit rate and they're bang on).

Next step was that I wrote two programs on my Amiga. One samples the parallel port (I called this program "scope") and the other (called "nrzi") takes the results from "scope" and decodes it back to the original data bits (and if the CRC is OK it also attempts to print the AX25 header).

I connect the NRZI digital output of the TNC's modem (3105) into the parallel port and "scope" samples this at any rate I choose.

If I play the DSP audio into the TNC and sample its modem's NRZI output, "nrzi" decodes it with no problem. But, if I look at the RS-232 output of the TNC itself it doesn't send the content of the packet to the computer.

The bit that really throws me is that if I connect the TNC to the audio from my radio and then run "scope" and "nrzi" on what the TNC hears, the "nrzi" program decodes "real" packets without any trouble too and this is using exactly the same sampling rates as I used to sample the audio from the DSP.

The fact that I can decode the NRZI output from the TNC modem eliminates several possible problems such as incorrect audio tones and the fact that

the "nrzi" program can decode "real" packets using the modem's NRZI output
suggests that the timing is also correct.

I'm lost. I know this is "back to basics" for most of you but has anyone
any idea where I can start looking for the problem?

73 de Pete
ve5va.qrp@usask.ca

From mwestfal@csci.csusb.edu Sat Jun 08 21:19:21 1996
Received: from silicon.csci.csusb.edu (silicon.csci.csusb.edu [139.182.38.1]) by
tapr.org (8.7.5/8.7.3/1.9) with SMTP id VAA09069 for <hfsig@tapr.org>; Sat, 8 Jun
1996 21:19:18 -0500 (CDT)
Received: by silicon.csci.csusb.edu (5.0/SMI-SVR4)
     id AA12396; Sat, 8 Jun 1996 19:33:41 +0800
From: mwestfal@csci.csusb.edu (Michael Westfall)
Received: by csci.csusb.edu id TAA20047; Sat, 8 Jun 1996 19:22:49 -0700 (PDT)
(8.7.1 Berkeley Sendmail)
Message-Id: <199606090222.TAA20047@csci.csusb.edu>
Subject: Re: [HFSIG:1201] Stuck at the beginning
To: hfsig@tapr.org
Date: Sat, 8 Jun 1996 19:22:48 -0700 (PDT)
In-Reply-To: <Pine.OSF.3.93.960608132158.5705A-100000@duke.usask.ca> from "Pete
Hardie" at Jun 8, 96 02:57:28 pm
Reply-To: mwestfal@csci.csusb.edu
X-Hi-Mom: Send more money!
Organization: The Hackers' Guild
X-Mailer: ELM [version 2.4 PL20]
Content-Type: text

> I'm just starting with DSP and thought (after creating some CW FIR
> filters) that I'd try to implement a 1200 baud packet TNC on the EZ-KIT
> Lite.
> I had assumed that implementing the modulation side of this was going to
> be easy but I've run into a major hurdle that I can't figure out.
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> Generating the NRZI data and modulating it as a continuous phase FSK audio
> signal was easy. I have set up the DSP program so that it reads KISS
> packets from the RS-232 port and then generates the FSK. However, if I put
> the audio from the Kit into either my KAM or Tiny-2 neither of them will
> decode a packet.

I would guess what you are forgetting is to add the HDLC frame flags and
bit-stuffing....

>
> I'm lost. I know this is "back to basics" for most of you but has anyone
> any idea where I can start looking for the problem?

I'm no expert, so I'll defer to someone else to explain further...
----
73 de Mike,   ax.25net: N6KUY@W6JBT.#SOCA.CA.USA.NA
                  amprnet: n6kuy@n6kuy.ampr.org [44.18.0.49]

internet :   mwestfal@csci.csusb.edu
web: http://orion.csci.csusb.edu:8080
"Linux: The Gates of Hell shall not prevail."
GCS/M { -d+ p+ c++ l u++ e+(*) m++(-) s/+ !n-(---) h-- !f g+ w+ t++ r-(--) y+ }

From hardie@duke.usask.ca Sat Jun 08 22:27:03 1996
Received: from duke.usask.ca (duke.usask.ca [128.233.3.13]) by tapr.org
(8.7.5/8.7.3/1.9) with ESMTP id WAA11050 for <hfsig@tapr.org>; Sat, 8 Jun 1996
22:27:00 -0500 (CDT)
Received: from localhost (hardie@localhost) by duke.usask.ca (8.7.3/8.7.3) with
SMTP id VAA08814 for <hfsig@tapr.org>; Sat, 8 Jun 1996 21:26:57 -0600 (CST)
Date: Sat, 8 Jun 1996 21:26:57 -0600 (CST)
From: Pete Hardie <hardie@duke.usask.ca>
To: hfsig@tapr.org
Subject: Re: [HFSIG:1202] Re: Stuck at the beginning
In-Reply-To: <199606090222.TAA20047@csci.csusb.edu>
Message-ID: <Pine.OSF.3.93.960608212140.22169A-100000@duke.usask.ca>
MIME-Version: 1.0
Content-Type: TEXT/PLAIN; charset=US-ASCII

On Sat, 8 Jun 1996, Michael Westfall wrote:

> I would guess what you are forgetting is to add the HDLC frame flags and
> bit-stuffing....

That's all in there Mike because:
(a) I know they are in the DSP program.
(b) My "nrzi" program which decodes the signals not only decodes mine
correctly from the DSP but it also correctly decodes real packets off the
air. So my DSP has to be generating the flags and bit-stuffing otherwise
the "nrzi" program wouldn't decode it.

Thanks for the reply.

73 de Pete
ve5va.qrp@usask.ca

From forrerj@ucs.orst.edu Sun Jun 09 01:00:05 1996
Received: from ucs.orst.edu (root@UCS.ORST.EDU [128.193.4.5]) by tapr.org
(8.7.5/8.7.3/1.9) with SMTP id BAA23096 for <hfsig@tapr.org>; Sun, 9 Jun 1996
01:00:03 -0500 (CDT)
Received: from p04.t0.monrotel.com by ucs.orst.edu;
(5.65v3.2/1.1.8.2/13Mar96-1233PM)
id AA05625; Sat, 8 Jun 1996 22:59:57 -0700
Message-Id: <1.5.4.16.19960609072233.38677264@ucs.orst.edu>
X-Sender: forrerj@ucs.orst.edu
X-Mailer: Windows Eudora Light Version 1.5.4 (16)
Mime-Version: 1.0
Content-Type: text/plain; charset="us-ascii"
Date: Sat, 08 Jun 1996 23:22:33 -0800
To: hfsig@tapr.org
From: Johan Forrer <forrerj@ucs.orst.edu>

Subject: Re: [HFSIG:1201] Stuck at the beginning

Hi Pete,

My two cent's worth:

>
>The fact that I can decode the NRZI output from the TNC modem eliminates
>several possible problems such as incorrect audio tones and the fact that
>the "nrzi" program can decode "real" packets using the modem's NRZI output
>suggests that the timing is also correct.
>

Seems like "nrzi" can decode both packets taken from your TNC's modem and your DSP signal source - so that looks like the format is OK.

Have you looked at the audio levels going to the real TNC when you try your DSP signal source? Perhaps they are overdriving the TNC. TNC's are quite sensitive to the level and the fact that you see DCD comes on may not mean too much. Perhaps some form of level adjustment may help. Otherwise, I would defininately listen to that audio with an amplified speaker or scope and determine whether is clean and without hum.

>I'm lost. I know this is "back to basics" for most of you but has anyone
>any idea where I can start looking for the problem?
>
>73 de Pete
>ve5va.qrp@usask.ca
>
>

Looks like you are well on your way.

--Johan

From hardie@duke.usask.ca Sun Jun 09 20:29:18 1996
Received: from duke.usask.ca (duke.usask.ca [128.233.3.13]) by tapr.org (8.7.5/8.7.3/1.9) with ESMTP id UAA00317 for <hfsig@tapr.org>; Sun, 9 Jun 1996 20:25:26 -0500 (CDT)
Received: from localhost (hardie@localhost) by duke.usask.ca (8.7.3/8.7.3) with SMTP id TAA30004 for <hfsig@tapr.org>; Sun, 9 Jun 1996 19:23:56 -0600 (CST)
Date: Sun, 9 Jun 1996 19:23:56 -0600 (CST)
From: Pete Hardie <hardie@duke.usask.ca>
To: hfsig@tapr.org
Subject: Re: [HFSIG:1204] Re: Stuck at the beginning
In-Reply-To: <1.5.4.16.19960609072233.38677264@ucs.orst.edu>
Message-ID: <Pine.OSF.3.93.960609191716.28968A-100000@duke.usask.ca>
MIME-Version: 1.0
Content-Type: TEXT/PLAIN; charset=US-ASCII

On Sun, 9 Jun 1996, Johan Forrer wrote:

> Seems like "nrzi" can decode both packets taken from your TNC's modem and
> your DSP signal source - so that looks like the format is OK.
>
> Have you looked at the audio levels going to the real TNC when you try your
> DSP signal source? Perhaps they are overdriving the TNC.

I have checked that out and it doesn't make any difference. I wasn't sure that it should because the signal that "nrzi" sees is what the modem has decoded from the audio input. If the audio were overdriving the modem then I would expect the program to see that as errors in the nrzi bitstream coming out of the modem but "nrzi" has no trouble decoding the bitstream.

> Otherwise, I would
> defininately listen to that audio with an amplified speaker or scope and
> determine whether is clean and without hum.

I have listened to the audio and it sounds exactly like the packets I hear on the air. I haven't got a scope but I'm going to try to borrow one to have a look at the nrzi bitstream from the TNC's modem. Perhaps there are subtle timing differences which don't show up due to the nature of my sampling program.

Thanks Johan.

Pete

From karn@qualcomm.com Mon Jun 10 19:40:27 1996
Received: from servo.qualcomm.com (servo.qualcomm.com [129.46.128.14]) by tapr.org (8.7.5/8.7.3/1.9) with ESMTP id TAA27043 for <hfsig@tapr.org>; Mon, 10 Jun 1996 19:40:17 -0500 (CDT)
Received: (from karn@localhost) by servo.qualcomm.com (8.7.5/1.0/8.7.2/1.9) id RAA29160; Mon, 10 Jun 1996 17:39:42 -0700 (PDT)
Date: Mon, 10 Jun 1996 17:39:42 -0700 (PDT)
From: Phil Karn <karn@qualcomm.com>
Message-Id: <199606110039.RAA29160@servo.qualcomm.com>
To: hfsig@tapr.org
In-reply-to: <1.5.4.16.19960608194055.3f8fec56@ucs.orst.edu> (message from Johan Forrer on Sat, 8 Jun 1996 13:35:25 -0500 (CDT))
Subject: Re: [HFSIG:1200] Re: Further thoughts on pulse shaping for HF

While the usual methods for extracting timing were designed for continuous streams of bits, I point out that just about all of our applications now involve packetized data of one sort or another. So that opens up some alternative methods for timing extraction, like a one-shot scheme based on a sync vector in the front of the packet.

That is, instead of continuously recovering clock from the data stream, you estimate it once during each packet and extrapolate it from there. This is easy to do by sliding a correlator over the packet -- when it hits its peak, you just count off the proper number of

samples per bit for the rest of the packet. Much easier.

With even cheap crystals being as good as they usually are, one-shot timing schemes work pretty well as long as the packet size is limited -- which it is anyway for many other reasons.

Too bad that one-shot schemes won't work for carrier recovery -- there's too much doppler, noise and/or other impairments for this to work, so you have to extract it as before with a Costas loop or something similar.

Phil

From forrerj@ucs.orst.edu Wed Jun 12 11:01:08 1996
Received: from ucs.orst.edu (forrerj@UCS.ORST.EDU [128.193.4.5]) by tapr.org (8.7.5/8.7.3/1.9) with SMTP id LAA29111 for <HFSIG@TAPR.ORG>; Wed, 12 Jun 1996 11:01:01 -0500 (CDT)
Received: by ucs.orst.edu; (5.65v3.2/1.1.8.2/13Mar96-1233PM)
     id AA05079; Wed, 12 Jun 1996 09:00:45 -0700
Date: Wed, 12 Jun 1996 09:00:45 -0700 (PDT)
From: Johan Forrer <forrerj@ucs.orst.edu>
To: HFSIG@tapr.org
Subject: Re: [HFSIG:1206] Re: Further thoughts on pulse shaping for HF
In-Reply-To: <65E6D933EE1@frl.orst.edu>
Message-Id: <Pine.OSF.3.91.960612084342.19054A-100000@ucs.orst.edu>
Mime-Version: 1.0
Content-Type: TEXT/PLAIN; charset=US-ASCII

Hi Phil,

>
>
> While the usual methods for extracting timing were designed for
> continuous streams of bits, I point out that just about all of our
> applications now involve packetized data of one sort or another. So
> that opens up some alternative methods for timing extraction, like a
> one-shot scheme based on a sync vector in the front of the packet.

It is true that nearly all communications nowadays have some form of header that allows for this kind of one-shot sync. Could you tell us a bit more about the sync vector that you are now using. I did follow some discussion a while ago about your explorations for a suitable sync vector.

I now have reasonable clock and data extraction working, but I am working on extending the code to look for a sync vector. Although transmission of the main data stream is DQPSK, I use ordinary DPSK for the sync vector. I'll do as you suggest - obtain and set the clock at that point it recognises the sync vector. My observations have been that there is a fraction of a bit's worth slippage across my packets - not too much to be of concern.

I also am considering using offset QPSK and was wondering; would the extended Costas loop as for QPSK work for offset QPSK, or would I

need to change it?

>
>
>That is, instead of continuously recovering clock from the data
> stream, you estimate it once during each packet and extrapolate it
> from there. This is easy to do by sliding a correlator over the packet
> -- when it hits its peak, you just count off the proper number of
> samples per bit for the rest of the packet. Much easier.
>
> With even cheap crystals being as good as they usually are, one-shot
> timing schemes work pretty well as long as the packet size is limited
> -- which it is anyway for many other reasons.
>
> Too bad that one-shot schemes won't work for carrier recovery -- there's
> too much doppler, noise and/or other impairments for this to work, so
> you have to extract it as before with a Costas loop or something similar.
>
> Phil
>
>

The idea here is to use a number of orthogonal channels for data, one
channel will be unmodulated just to extract doppler.

The project has been a lot of fun but sure burns time and effort.
However, I'm learning a lot too.

--Johan

From karn@qualcomm.com Wed Jun 12 23:21:21 1996
Received: from servo.qualcomm.com (servo.qualcomm.com [129.46.128.14]) by tapr.org (8.7.5/8.7.3/1.9) with ESMTP id XAA27501 for <hfsig@tapr.org>; Wed, 12 Jun 1996 23:21:19 -0500 (CDT)
Received: (from karn@localhost) by servo.qualcomm.com (8.7.5/1.0/8.7.2/1.9) id VAA01103; Wed, 12 Jun 1996 21:20:42 -0700 (PDT)
Date: Wed, 12 Jun 1996 21:20:42 -0700 (PDT)
From: Phil Karn <karn@qualcomm.com>
Message-Id: <199606130420.VAA01103@servo.qualcomm.com>
To: hfsig@tapr.org
In-reply-to: <Pine.OSF.3.91.960612084342.19054A-100000@ucs.orst.edu> (message from Johan Forrer on Wed, 12 Jun 1996 11:09:06 -0500 (CDT))
Subject: Re: [HFSIG:1207] Re: Further thoughts on pulse shaping for HF

>header that allows for this kind of one-shot sync. Could you tell us a
>bit more about the sync vector that you are now using. I did follow some

Well, I haven't settled on a final vector yet, but at the moment I'm
using a 26-bit vector constructed from the 13-bit Barker code (the
longest-known binary Barker code) with each bit Manchester encoded.
I.e., a "1" becomes "10" and a "0" becomes "01".

This does several things. It adds energy to the total vector, making

it easier to spot reliably, and it also adds two negative sidelobes
right next to the main positive lobe. In radar where you don't have
polarity information and are forced to look at signal envelopes, this
would be a big drawback. But this is actually an advantage to me since

I already have signal polarity from the carrier preamble, and the negative
sidelobes "sharpen" the main lobe, making it less likely for me to lock
one (or more) samples off the center of the main lobe.

I've simulated this vector pretty heavily and it seems adequate at the
nominal modem Eb/N0 of 3dB, but I wouldn't mind a little more margin.

Phil

From karn@qualcomm.com Wed Jun 12 23:30:57 1996
Received: from servo.qualcomm.com (servo.qualcomm.com [129.46.128.14]) by tapr.org
(8.7.5/8.7.3/1.9) with ESMTP id XAA27681 for <hfsig@tapr.org>; Wed, 12 Jun 1996
23:30:53 -0500 (CDT)
Received: (from karn@localhost) by servo.qualcomm.com (8.7.5/1.0/8.7.2/1.9) id
VAA01120; Wed, 12 Jun 1996 21:30:22 -0700 (PDT)
Date: Wed, 12 Jun 1996 21:30:22 -0700 (PDT)
From: Phil Karn <karn@qualcomm.com>
Message-Id: <199606130430.VAA01120@servo.qualcomm.com>
To: hfsig@tapr.org
In-reply-to: <Pine.OSF.3.91.960612084342.19054A-100000@ucs.orst.edu> (message from
Johan Forrer on Wed, 12 Jun 1996 11:09:06 -0500 (CDT))
Subject: Re: [HFSIG:1207] Re: Further thoughts on pulse shaping for HF

>I also am considering using offset QPSK and was wondering; would the
>extended Costas loop as for QPSK work for offset QPSK, or would I
>need to change it?

Yes, it works. The scheme I've been using is to sample both I&Q twice
per symbol, once at the center of the I symbol and again at the center
of the Q symbol. Two of these samples obviously become the recovered
data symbols. The out-of-phase samples (e.g., the one of the Q-channel
taken at the I-channel sampling instant) has its sign flipped by the
polarity of the current in-phase sample. Then the polarity corrected
out-of-phase Q-channel samples are subtracted from the polarity
corrected out-of-phase I-channel samples, producing the phase
residuals that my loop attempts to minimize in a least-squares
fashion.

This is basically the classic decision-directed 4-phase costas loop,
implemented in a sampled system.

I've been looking more closely at your exact question over the past
day or two. With my current scheme it *seems* that with SQPSK (but not
plain QPSK) there exist data patterns that produce no phase detector
output even when there is a carrier phase error. If you number the

four phase states 1-2-3-4, imagine an encoded phase sequence that goes 1-2-3-4-1-2-3-4 (or 4-3-2-1-4-3-2-1). (This corresponds roughly to an unmodulated carrier offset above or below the real carrier). In this case, all of the out-of-phase symbols are in the process of changing when they're sampled by the in-phase symbol, leaving no residual even when there's a carrier phase error.

The scheme still seems to work, at least for the random-like data produced by the convolutional coder and interleaver, but this phenomenon still bothers me. If nothing else, it means that the gain of the "loop" (and the speed at which it converges) depends on the signal pattern as well as the amplitude of the signal. Although this doesn't actually affect the error performance of the modem (since I operate in an a-posteriori batch mode where I repeatedly crunch each block of symbols until I find the steady-state maximum likelihood carrier phase and frequency), it does affect the cost in CPU cycles.

More when I figure this all out for myself.

Phil

From karn@qualcomm.com Wed Jun 12 23:34:53 1996
Received: from servo.qualcomm.com (servo.qualcomm.com [129.46.128.14]) by tapr.org (8.7.5/8.7.3/1.9) with ESMTP id XAA27983 for <hfsig@tapr.org>; Wed, 12 Jun 1996 23:34:50 -0500 (CDT)
Received: (from karn@localhost) by servo.qualcomm.com (8.7.5/1.0/8.7.2/1.9) id VAA01148; Wed, 12 Jun 1996 21:34:19 -0700 (PDT)
Date: Wed, 12 Jun 1996 21:34:19 -0700 (PDT)
From: Phil Karn <karn@qualcomm.com>
Message-Id: <199606130434.VAA01148@servo.qualcomm.com>
To: hfsig@tapr.org
In-reply-to: <Pine.OSF.3.91.960612084342.19054A-100000@ucs.orst.edu> (message from Johan Forrer on Wed, 12 Jun 1996 11:09:06 -0500 (CDT))
Subject: Re: [HFSIG:1207] Re: Further thoughts on pulse shaping for HF

>The idea here is to use a number of orthogonal channels for data, one
>channel will be unmodulated just to extract doppler.

How well does the carrier reference channel work in practice, given multipath? Can you really just lock a loop to it and derive carrier phase for all your other channels, eliminating all the squaring (or worse) losses inherent in suppressed carrier systems?

Phil

From forrerj@ucs.orst.edu Thu Jun 13 11:38:37 1996
Received: from ucs.orst.edu (root@UCS.ORST.EDU [128.193.4.5]) by tapr.org (8.7.5/8.7.3/1.9) with SMTP id LAA27391 for <hfsig@tapr.org>; Thu, 13 Jun 1996 11:38:34 -0500 (CDT)
Received: from p00.t0.monrotel.com by ucs.orst.edu;
(5.65v3.2/1.1.8.2/13Mar96-1233PM)
     id AA14059; Thu, 13 Jun 1996 09:38:27 -0700
Message-Id: <1.5.4.16.19960613180144.3ff78574@ucs.orst.edu>

X-Sender: forrerj@ucs.orst.edu
X-Mailer: Windows Eudora Light Version 1.5.4 (16)
Mime-Version: 1.0
Content-Type: text/plain; charset="us-ascii"
Date: Thu, 13 Jun 1996 10:01:44 -0800
To: hfsig@tapr.org
From: Johan Forrer <forrerj@ucs.orst.edu>
Subject: Re: [HFSIG:1210] Re: Further thoughts on pulse shaping for HF

Phil,

At 11:49 PM 6/12/96 -0500, you wrote:
>>The idea here is to use a number of orthogonal channels for data, one
>>channel will be unmodulated just to extract doppler.
>
>How well does the carrier reference channel work in practice, given
>multipath? Can you really just lock a loop to it and derive carrier
>phase for all your other channels, eliminating all the squaring (or
>worse) losses inherent in suppressed carrier systems?
>
>Phil
>
>

That's a good question. Perhaps someone like Hakan, who are more familiar with MIL-STD 188 or STANAG modems could tell us more about that. I have seen the idea used for doppler compensation in both the 39 and 16 tone modems for this purpose. Perhaps to deal with a kind of doppler that affects the block of frequencies as a whole instead of only selective channels. As to how valid or effective that is, I really don't know.

Someone care to comment?

--Johan

From forrerj@ucs.orst.edu Thu Jun 13 11:38:40 1996
Received: from ucs.orst.edu (root@UCS.ORST.EDU [128.193.4.5]) by tapr.org (8.7.5/8.7.3/1.9) with SMTP id LAA27392 for <hfsig@tapr.org>; Thu, 13 Jun 1996 11:38:37 -0500 (CDT)
Received: from p00.t0.monrotel.com by ucs.orst.edu;
(5.65v3.2/1.1.8.2/13Mar96-1233PM)
     id AA04085; Thu, 13 Jun 1996 09:38:22 -0700
Message-Id: <1.5.4.16.19960613180138.3ff7efd0@ucs.orst.edu>
X-Sender: forrerj@ucs.orst.edu
X-Mailer: Windows Eudora Light Version 1.5.4 (16)
Mime-Version: 1.0
Content-Type: text/plain; charset="us-ascii"
Date: Thu, 13 Jun 1996 10:01:38 -0800
To: hfsig@tapr.org
From: Johan Forrer <forrerj@ucs.orst.edu>
Subject: Re: [HFSIG:1209] Re: Further thoughts on pulse shaping for HF

Hi Phil,

At 11:49 PM 6/12/96 -0500, you wrote:
>>I also am considering using offset QPSK and was wondering; would the
>>extended Costas loop as for QPSK work for offset QPSK, or would I
>>need to change it?
>
>Yes, it works. The scheme I've been using is to sample both I&Q twice
>per symbol, once at the center of the I symbol and again at the center
>of the Q symbol. Two of these samples obviously become the recovered
>data symbols. The out-of-phase samples (e.g., the one of the Q-channel
>taken at the I-channel sampling instant) has its sign flipped by the
>polarity of the current in-phase sample. Then the polarity corrected
>out-of-phase Q-channel samples are subtracted from the polarity
>corrected out-of-phase I-channel samples, producing the phase
>residuals that my loop attempts to minimize in a least-squares
>fashion.

Thanks - good to know that.

>
>This is basically the classic decision-directed 4-phase costas loop,
>implemented in a sampled system.
>
>I've been looking more closely at your exact question over the past
>day or two. With my current scheme it *seems* that with SQPSK (but not
>plain QPSK) there exist data patterns that produce no phase detector
>output even when there is a carrier phase error. If you number the
>four phase states 1-2-3-4, imagine an encoded phase sequence that goes
>1-2-3-4-1-2-3-4 (or 4-3-2-1-4-3-2-1). (This corresponds roughly to an
>unmodulated carrier offset above or below the real carrier). In this
>case, all of the out-of-phase symbols are in the process of changing
>when they're sampled by the in-phase symbol, leaving no residual even
>when there's a carrier phase error.

I follow - interesting that SQPSK has that anomaly. I was going to ask
whether you are using a scrambler as that would reduce the recurrence of
such a pattern, but your convolutional coding and interleaver will probably
have the same effect to randomize the pattern.

>
>The scheme still seems to work, at least for the random-like data
>produced by the convolutional coder and interleaver, but this
>phenomenon still bothers me. If nothing else, it means that the gain
>of the "loop" (and the speed at which it converges) depends on the
>signal pattern as well as the amplitude of the signal. Although this
>doesn't actually affect the error performance of the modem (since I
>operate in an a-posteriori batch mode where I repeatedly crunch each
>block of symbols until I find the steady-state maximum likelihood
>carrier phase and frequency), it does affect the cost in CPU cycles.

I played with a decision-directed scheme for a while that, at the middle of
the symbol, found the closest constellation point, and then determines

whether the received constellation needed to be rotated clockwise or
anti-clockwise. When the set of constellation points were {(1 0), (0 -1),
(-1 0), (0 -1)}, then either I or Q needed to be zero. One could then use
this non-zero offset as an (magnitude, direction) error estimate to be used
for loop control. This worked wonderfully until I started using RC pulse
shaping - the zero-crossing transitions for the RC pulses just played havoc.
At that point I switched to the extended Costas loop that took care of it,
however, as you said in an earlier posting, that type of loop drives I and Q
maximal and in my case, the transmitted and received constallations are
offset by a constant 45 degrees. That does not bother differential coding as
long as it is a stable lock, which is the case. I find that I can start the
demodulator midst of a random data sequence, obtain and hold lock - even for
33 second packets.

>
>More when I figure this all out for myself.
>
>Phil
>
>

Let us know the outcome - I'm curious.

--Johan

From zs6awk@global.co.za Fri Jun 14 16:51:36 1996
Received: from lin01.global.co.za (lin01.global.co.za [196.3.164.2]) by tapr.org
(8.7.5/8.7.3/1.9) with ESMTP id QAA03934 for <hfsig@tapr.org>; Fri, 14 Jun 1996
16:50:03 -0500 (CDT)
Received: from mail.global.co.za ([196.3.168.58]) by lin01.global.co.za
(8.7.3/8.7.3) with SMTP id XAA27977 for <hfsig@tapr.org>; Fri, 14 Jun 1996
23:48:12 -0200 (GMT)
Message-Id: <199606150148.XAA27977@lin01.global.co.za>
X-Sender: zs6awk@mail.global.co.za (Unverified)
X-Mailer: Windows Eudora Version 1.4.4
Mime-Version: 1.0
Content-Type: text/plain; charset="us-ascii"
Date: Fri, 14 Jun 1996 23:49:42 +0200
To: hfsig@tapr.org
From: zs6awk@global.co.za (Danie Brynard)
Subject: re pos of filter vs nyquist

I have been playing around with simulations to see if the position of a FIR
filter is important relative to the Nyquist frequency. Thus if I have a BPF
of say arbitrary passband and stopband does it matter where I place this
filter in the frequency range 0Hz to Fnyquist Hz ? (Fnyq=Fs/2)

According to my simulations it does not matter but I am not convinced. In
the analog world the rule is: the higher the freqency the more difficult it
is to make the same BPF.

Is wordlength or calculation precision perhaps the answer ? Is the number of
cycles per unit time of the wanted signal perhaps the answer ?

Could somebody give me a 'gut-feel' answer ? In other words a layman explanation why it does/does not matter where one puts the filter.

I hope I explained myself well enough. Looking forward to all the interesting replies :-)

73 danie zs6awk@global.co.za

From hakan.bergzen@enator.se Sat Jun 15 10:40:42 1996
Received: from gk.enator.se (ns.enator.se [147.13.200.1]) by tapr.org
(8.7.5/8.7.3/1.9) with ESMTP id KAA18040 for <hfsig@tapr.org>; Sat, 15 Jun 1996
10:40:31 -0500 (CDT)
Received: from janus.vxo.telub.se ([147.13.8.25]) by gk.gk.enator.se with SMTP id
<35746>; Sat, 15 Jun 1996 17:40:37 +0100
Received: from noak.vxo.enator.se by janus.vxo.telub.se with SMTP (PP)
          id <11592-0@janus.vxo.telub.se>; Sat, 15 Jun 1996 17:35:10 +0200
Received: by noak with Microsoft Mail id <31CDE3D2@noak>;
          Sat, 15 Jun 96 17:39:46 +02
From: "Bergzen Hakan, KARL" <hakan.bergzen@enator.se>
To: hfsig <hfsig@tapr.org>
Subject: Re: Further thoughts on pulse shaping for HF
Date: Sat, 15 Jun 1996 16:39:00 +0100
Message-ID: <31CDE3D2@noak>
Return-Receipt-To: HABE <hakan.bergzen@enator.se>
Encoding: 67 TEXT
X-Mailer: Microsoft Mail V3.0
MIME-Version: 1.0
Content-Type: Text/plain; charset="US-ASCII"
Content-Transfer-Encoding: 7bit

Johan,

I have not been able to monitor this group for some time, just trying to catch up, when I saw your reference to me. Even though I doubt I would have some information which not you and Phil already knows (I am very impressed of both of you being able to give the rest of us elaborate responses to our various questions, comments and ideas). This probably also goes for several silent readers of this group.

>>>The idea here is to use a number of orthogonal channels for data, one
>>>channel will be unmodulated just to extract doppler.
>>
>>How well does the carrier reference channel work in practice, given
>>multipath? Can you really just lock a loop to it and derive carrier
>>phase for all your other channels, eliminating all the squaring (or
>>worse) losses inherent in suppressed carrier systems?
>>
>That's a good question. Perhaps someone like Hakan, who are more familiar
>with MIL-STD 188 or STANAG modems could tell us more about that. I have
seen
>the idea used for doppler compensation in both the 39 and 16 tone modems

for
>this purpose. Perhaps to deal with a kind of doppler that affects the block
>of frequencies as a whole instead of only selective channels. As to how
>valid or effective that is, I really don't know.

The parallel tone modes within MIL-STD-188-110A use a special doppler correction tone, which will run continuously. It is the lowest tone in both modes (605 Hz for the 16-tone mode and 393.75 Hz for the 39-tone mode). It is only used for doppler correction, thus enabling the DSP to adjust the frequency offset for all data tones up or down correspondingly before symbol demodulation.

Both modes use BDPSK and QDPSK (depending on the data rate selected). At the lower data rates DPSK and in-band diversity combining are used. The 39-tone mode also use interleavers. Since only differential encoding is used the symbol decoding problem is more or less eliminated (I dont think it is possible to extract a carrier phase for one tone and use it for any other tone).

The protocol uses an initial preamble. The 39-tone mode use a 3-part preamble of different data tones all-in-all 23 signal elements (An extended preamble uses 97 signal elements). The data to be sent is divided into blocks. Each data block starts with a block sync. The block length depends on interleaver depth and data rate selected. A transmission is ended with an End-of-message sequence.

I think it would be a good idea to include a superb feature defined in the serial tone mode of the MIL-STD-188-110A, namely the Autobaud function. Within the initial preamble (and the re-occuring preamble enabling re-sync) the data rate and interleaver setting is encoded. This enables a receiving modem to automatically make the correct parameter setting and receive the message. The transmitting modem can by itself decide upon the most suitable data rate/interleaver depth.This makes it e.g. possible to easily construct adaptive ARQ schemes.

The trend in Europe and the U.S. seems to go from parallel tone modems to serial tone modems. They are more tricky to construct (including adaptive channel equalizers) but they show better performance because of their better utilization of the available channel (a sort of narrow-band SS, really). Though recent Australian studies (DSTO) using parallel tone modems together with trellis coding claims to perform better than serial tone modems.

For what it's worth...

Hakan   (hakan.bergzen@enator.se)

From JSANFORD@INFI.NET Sat Jun 15 20:05:03 1996
Received: from mh004.infi.net (mailhost.infi.net [205.219.238.95]) by tapr.org (8.7.5/8.7.3/1.9) with ESMTP id UAA07794 for <hfsig@tapr.org>; Sat, 15 Jun 1996 20:04:48 -0500 (CDT)
Received: from pa9dsp3.orf.infi.net by mh004.infi.net with SMTP
     (Infinet-S-3.3) id VAA28286; Sat, 15 Jun 1996 21:04:17 -0400 (EDT)
Message-Id: <199606160104.VAA28286@mh004.infi.net>

X-Sender: jsanford@infi.net
X-Mailer: Windows Eudora Light Version 1.5.2
Mime-Version: 1.0
Content-Type: text/plain; charset="us-ascii"
Date: Sat, 15 Jun 1996 21:04:37 -0400
To: hfsig@tapr.org
From: Jim Sanford <JSANFORD@INFI.NET>
Subject: Re: [HFSIG:1190] Re: QPSK BANDWIDTH ?

Phil
Forgive the long delay . .. the cost of my job which takes me
to sea . . .

Your comment about filters is part of why I'm hoping to do the thing
with simple chips and do ALL filtering in DSP. Of course, Ulrich Rhode
will remind me that a "ROOFING" filter would improve s/n in the IF . . .

Goal is acceptable performance at low cost and ease of duplication. I'm
hoping to turn this project into a kit for our club . . .

At 03:22 PM 6/4/96 -0500, you wrote:
>>   1. For what platform are you doing the OQPSK
>>satellite modem? I have 2 dsp-12's and would love to
>>beta test . . .
>
>The platform is a generic 486 or Pentium PC running my KA9Q NOS TCP/IP
>package. The modem will be integrated in as an interface driver for a
>SoundBlaster 16 card *without* any special DSP chips.
>
>Your work with the 'digital friendly' receiver sounds very
>interesting! I do hope that whatever you come up with can be easily
>replicated by the average amateur -- in my experience, that's where
>many promising amateur hardware projects fall down. It's the main
I absolutely agree...

>reason I've become so interested in using mass-market PC hardware to
>the greatest possible extent, doing everything else in software. But
>there are definite limits to the available hardware (particularly in
>the radios) so your work is most welcome. I agree, this stuff ought
>not to be as expensive and complicated as it seems to be.
I've been somewhat against using stuff in a PC because I hate the thought
of tying up a PC for a single function, but you've got a GREAT point
about the cost of mass market hardware vs anything for the much smaller
ham market. And, as costs of PC's drop . . . .

I'm already dedicating a 486/overdrive to the pacsats, so, what's another
cheap 486 on the network . . . <grin>

Thanks again for all you've done....73, Jim
wb4gcs

>The guy behind the Fox-Tango table, being used to selling very sharp
>and narrow crystal filters for HF contest and DX use, couldn't figure

```
>out why in the world I would want to make my rig go *wider*. Ah, the
>"narrowband mindset" seems pretty well entrenched. :-)
>

From zs6awk@global.co.za  Sun Jun 16 14:55:40 1996
Received: from lin01.global.co.za (lin01.global.co.za [196.3.164.2]) by tapr.org
(8.7.5/8.7.3/1.9) with ESMTP id OAA25148 for <hfsig@tapr.org>; Sun, 16 Jun 1996
14:55:30 -0500 (CDT)
Received: from anx_80.global.co.za (anx_80.global.co.za [196.3.164.130]) by
lin01.global.co.za (8.7.3/8.7.3) with SMTP id VAA03929 for <hfsig@tapr.org>; Sun,
16 Jun 1996 21:53:37 -0200 (GMT)
Message-Id: <199606162353.VAA03929@lin01.global.co.za>
X-Sender: zs6awk@mail.global.co.za (Unverified)
X-Mailer: Windows Eudora Version 1.4.4
Mime-Version: 1.0
Content-Type: text/plain; charset="us-ascii"
Date: Sun, 16 Jun 1996 21:55:02 +0200
To: hfsig@tapr.org
From: zs6awk@global.co.za (Danie Brynard)
Subject: Re: [HFSIG:868] Re: DSP Tools

Phil is there perhaps a book that you can suggest if one wants to read more
about the inner working of the 486 and Pentuim and also on real-time
programming of it in assembler ? Something perhaps like 'asm programming on
the Pentuim for beginners' :-)

I am hooked onto the Motorola DSP56xxx family but is interested in your ideas..

danie zs6awk
zs6awk@global.co.za

>At 06:22 PM 2/8/96 -0600, you wrote:
>
>>As for DSP development platforms.  Both TI (www.ti.com) and Motorola
>>(www.motorola.com) have inexpensive DSP Starter Kits (DSK) (TI) or
>>Evaluation Modules (EVM) (Motorola).  The TI C5X DSK (p/n TMDS3200051) costs
>>$99.00 and the Motorola EVM560002 costs $149.00.  I own the later and have
>>the former on order.
>
>Don't forget that you can do serious DSP on general purpose computers without
>a DSP board. Pentiums and the faster 486s rival many dedicated DSPs, especially
>on floating point computations.
>
>Phil
>
>

From karn@unix.ka9q.ampr.org Sun Jun 16 16:51:29 1996
Received: from unix.ka9q.ampr.org (karn@unix.ka9q.ampr.org [129.46.90.35]) by
tapr.org (8.7.5/8.7.3/1.9) with ESMTP id QAA29942 for <hfsig@tapr.org>; Sun, 16
Jun 1996 16:51:25 -0500 (CDT)
Received: (from karn@localhost) by unix.ka9q.ampr.org (8.7.3/8.6.12) id OAA15153;
Sun, 16 Jun 1996 14:51:19 -0700 (PDT)
```

Date: Sun, 16 Jun 1996 14:51:19 -0700 (PDT)
Message-Id: <199606162151.OAA15153@unix.ka9q.ampr.org>
From: Phil Karn <karn@unix.ka9q.ampr.org>
To: hfsig@tapr.org
In-reply-to: <199606162353.VAA03929@lin01.global.co.za> (zs6awk@global.co.za)
Subject: Re: [HFSIG:1216] Re: DSP Tools
Reply-To: karn@qualcomm.com

Well, the standard Intel programmer's references are pretty good. They tell you all you really need to know to optimize DSP code on the Pentium, such as instruction clock counts and considerations on how to keep the pipelines full.

Phil

From Robert.Glassey@nmp.nokia.com Mon Jun 17 04:27:27 1996
Received: from noknic.nokia.com (noknic.nokia.com [131.228.6.10]) by tapr.org (8.7.5/8.7.3/1.9) with SMTP id EAA02862 for <hfsig@tapr.org>; Mon, 17 Jun 1996 04:27:12 -0500 (CDT)
From: Robert.Glassey@nmp.nokia.com
Received: from samail01.nmp.nokia.com (samail01.nmp.nokia.com [131.228.240.6]) by noknic.nokia.com (8.6.9/8.6.9) with ESMTP id MAA10028 for <hfsig@tapr.org>; Mon, 17 Jun 1996 12:26:07 +0300
Received: from by samail01.nmp.nokia.com with SMTP
     (1.37.109.16/16.2) id AA110443452; Mon, 17 Jun 1996 12:24:12 +0300
X-Openmail-Hops: 2
Date: Mon, 17 Jun 96 10:17:35 +0100
Message-Id: <H000029201ddaff7@MHS>
Subject: BTL performance tests
Mime-Version: 1.0
To: hfsig@tapr.org
Content-Type: text/plain; charset=ISO-8859-1; name="BTL"
Content-Transfer-Encoding: 7bit

Hi all,

Over the weekend I did a few performance tests on my PC-Sound Blaster based DSP RTTY modem (BTL), and compared it with my PK232MBX. Here's the results. Any coments welcome.

	PK232MBX	BTL Ver 0.2
1 Noise performance: Eb/No for BER=10^-2	15dB	12dB
2 Adjacent channel rejection (+500Hz 98 baud RTTY)		
2a For 3dB rise in 10^-2 Eb/No:		
Square pulse interferer	9dB	30dB
Raised cosine pulse interferer	-	32dB
2b For clean wanted signal, 10^-2 BER:		
Square pulse interferer	16dB	32dB

Raised cosine pulse interferer - 35dB

Observations:

1. The BER taken includes errors due to both sync errors and data bit errors. For both modems relative contributions were about 50-50. If a bit synchronous mode were used rather than asynchronous baudot, BER would be half. My measurements show this is equivelent to a 1dB improvement in Eb/No for both modems.

2. Two different interferer types were used. Both were phase continous FSK, 98 baud baudot, centred 500Hz above the wanted signal (wanted 1275/1445, interferer 1775/1945) Standard RTTY has a square pulse shape resulting in high adjacent channel power. (-30dB @ 500Hz) This limited measurements (and is a real on air limit too) However I tried shaping the interferering FSK pulses with raised cosine edges (shaped 64 of 82 samples). ACP was reduced to -45dB @ 500Hz. Using this interferer, the true performance of the modem could be determined. This shaping reduced the interference RMS level to 72% and this was taken into account.

   The minimum signal level without intefernce was found to be 1 ADC step peak to peak, with dithering from the noise of almost 3 ADC steps peak-peak. (signals were calculated in 16 bits then quantised to 8 bits to allow noise dithering). At this level performance was still 12dB Eb/No @ BER 10^-2. This minimum signal level was 44dB below the inteference level used above. Thus quantisation was not a limiting factor in the above measurements. Lower levels were possible but Eb/No was worsened and quantisation boundries became significant.

Measurement Technique:

1. Eb/No measurements

Sampling rate was 8KHz

The signal was 45 baud Baudot, phase continuous AFSK, 1275Hz space and 1445Hz mark (170Hz shift). RMS to peak ratio = 0.71

The noise was generated by a 24bit maximum length PRN generator (poly=0x8D0000), doing 8 shifts per sample to get an 8 bit uniform random noise source. RMS to peak ratio of a uniform random noise source = 0.58

I have assumed a uniform distribution is OK since after
bandpass filtering and integration the noise distribution
aproaches gausian. (roughly speaking, the noise is averaged
over the bit period, ie 178 samples, making the noise
gausian by summation)

For Eb/No calculations I have used the RMS levels at the
8kHz sample rate.

Peak-peak levels of signal=45 and noise=128 becomes 12db
Eb/No thus:

```
Eb/No =  20*log10 ((45/2 * 0.71) / (128/2 * 0.58))
         + 10*log10(4000 Hz) - 10*log10(45 BPS)
      =  12.2 dB
```

2. Adjacent channel measurements

The signals are described above.

The interferer baud rate of 98 baud was chosen to try to
ovoid synchronisation between the wanted and interferer
so errors would appear more random, hopefully giving a
more accurate BER. However it does make the interferer
wider, but pulse shaping dealt with that. This may not
be the best method.

Results are given for two different cases. 1. For a
3dB rise in the Eb/No required for 10^-2 BER, and
2. The interference level that gives a BER of 10^-2
when the wanted signal is clean. These figures show
respectivly the degradation in weak signal performance,
and the maximum tolerable adjacent channel signal level
when the wanted signal is clean.

So, 12dB Eb/No doesn't sound all that hot, does it? But is this what
should be expected for FSK without ECC? Eb/No would be 11dB if a bit
synchronous mode was used. These are for a BER of 10^-2.

Cheers,

Rob

From karn@unix.ka9q.ampr.org Mon Jun 17 05:16:20 1996
Received: from unix.ka9q.ampr.org (karn@unix.ka9q.ampr.org [129.46.90.35]) by
tapr.org (8.7.5/8.7.3/1.9) with ESMTP id FAA04098 for <hfsig@tapr.org>; Mon, 17

Jun 1996 05:16:11 -0500 (CDT)
Received: (from karn@localhost) by unix.ka9q.ampr.org (8.7.3/8.6.12) id DAA00188;
Mon, 17 Jun 1996 03:16:05 -0700 (PDT)
Date: Mon, 17 Jun 1996 03:16:05 -0700 (PDT)
Message-Id: <199606171016.DAA00188@unix.ka9q.ampr.org>
From: Phil Karn <karn@unix.ka9q.ampr.org>
To: hfsig@tapr.org
In-reply-to: <199606130430.VAA01120@servo.qualcomm.com> (karn)
Subject: Re: [HFSIG:1209] Re: Further thoughts on pulse shaping for HF
Reply-To: karn@qualcomm.com

FYI, more about offset QPSK vs conventional QPSK.

It looks as though the advantages of staggered (offset) QPSK are outweighted by the disadvantages, so I've switched to straight QPSK.

There are two main advantages of offset QPSK: 1) less envelope droop when the signal is bandlimited, which turns into less sideband regrowth when the signal is nonlinearly amplified, and 2) less crosstalk between channels when the carrier phase reference is inaccurate.

I originally chose SQPSK for reason #2; I don't really care too much about #1 since I am going to operate in a power-controlled regime where I assume there'll be a fair amount of transmit headroom most of the time, hence plenty of linearity. And even if there were sideband regrowth it wouldn't be as important a source of interference since I'm already operating at very low S/N ratios thanks to power control and coding.

But it turns out that the crosstalk resistance of SQPSK is largely mitigated by greatly increased pattern-sensitive jitter in the carrier recovery system. In other words, although SQPSK has increased carrier phase jitter tolerance, it inherently increases the jitter of the recovered carrier which squanders the improved performance.

I couldn't find a precise quantitative formulation for this, but from what I saw it looks like any gains SQPSK might have (especially at the very low Es/N0s I'm using) are small or even negative. And the last straw was that SQPSK is harder to implement than straight QPSK. So I've switched to straight QPSK.

Phil

From karn@unix.ka9q.ampr.org Mon Jun 17 05:47:56 1996
Received: from unix.ka9q.ampr.org (karn@unix.ka9q.ampr.org [129.46.90.35]) by
tapr.org (8.7.5/8.7.3/1.9) with ESMTP id FAA05064 for <hfsig@tapr.org>; Mon, 17
Jun 1996 05:47:51 -0500 (CDT)
Received: (from karn@localhost) by unix.ka9q.ampr.org (8.7.3/8.6.12) id DAA00230;
Mon, 17 Jun 1996 03:47:44 -0700 (PDT)
Date: Mon, 17 Jun 1996 03:47:44 -0700 (PDT)

Message-Id: <199606171047.DAA00230@unix.ka9q.ampr.org>
From: Phil Karn <karn@unix.ka9q.ampr.org>
To: hfsig@tapr.org
In-reply-to: <1.5.4.16.19960613180138.3ff7efd0@ucs.orst.edu> (message from Johan Forrer on Thu, 13 Jun 1996 11:58:25 -0500 (CDT))
Subject: Re: [HFSIG:1212] Re: Further thoughts on pulse shaping for HF
Reply-To: karn@qualcomm.com

>I follow - interesting that SQPSK has that anomaly. I was going to ask
>whether you are using a scrambler as that would reduce the recurrence of
>such a pattern, but your convolutional coding and interleaver will probably
>have the same effect to randomize the pattern.

Re the anomaly I described (certain data patterns killing the SQPSK phase detector) I'm now not exactly sure that this is real. But it *does* seem from reading the books that there is a pattern-sensitive phase jitter in SQPSK that's not present in plain QPSK, and the problem gets worse when you tightly filter the signal. So that's why I've switched from SQPSK to plain QPSK. Simplified the code, too.

Re scramblers, no, I'm not scrambling. Yes, the convolutional coding and interleaving does a pretty good job. It certainly sounds noiselike except when I send a big block of zeroes, which come out as a tone. But the loop can track this fine anyway (remember I'm doing one-shot symbol timing up front).

Re your discussion of decision-directed carrier recovery, it works great when the Es/N0 is high enough to work without coding. But at the Es/N0 ratios I'm using with my coding, there are plenty of errors in the decisions because they're made by simple slicing, without benefit of any error correction. So many of the phase detector outputs are wrong (though not most, since the error rate is still less than 50%). This adds noise to the feedback loop over and above the noise in the input signal. In the literature this is called "squaring loss" (although strictly speaking that applies only to BPSK -- in QPSK it should be called "4th power loss"). Squaring loss is a function of loop design, Es/N0 and of the loop signal-to-noise ratio.

Squaring loss forces you to use a much narrower loop filter than you'd otherwise use in order to maintain an acceptably high S/N ratio within the loop. That means you get pretty sensitive to doppler.

Another good illustration of how trying to conserve bandwidth (e.g., by using QPSK instead of BPSK in the first place) often ends up forcing you to use more power per bit even when it shouldn't -- ideal QPSK and BPSK (with perfect carrier recovery) have exactly the same Eb/N0 requirements. I'm still having a hard time getting to within 1 dB of the theoretical code performance in my QPSK modem.

Phil

From LANIER.R.A-@smtpgty.bwi.wec.com Mon Jun 17 07:25:59 1996
Received: from tron.bwi.wec.com (tron.bwi.wec.com [129.228.4.1]) by tapr.org

(8.7.5/8.7.3/1.9) with SMTP id HAA09672; Mon, 17 Jun 1996 07:25:51 -0500 (CDT)
Received: from smtpgty.bwi.wec.com by tron.bwi.wec.com;
(5.65/1.1.8.2/31May95-0229PM)
id AA24688; Mon, 17 Jun 1996 07:46:00 -0400
Received: from ccMail by smtpgty.bwi.wec.com
(IMA Internet Exchange 2.0 Enterprise) id 1C54EE50; Mon, 17 Jun 96 08:26:13 -0400
Mime-Version: 1.0
Date: Mon, 17 Jun 1996 08:16:07 -0400
Message-Id: <1C54EE50.1858@smtpgty.bwi.wec.com>
From: LANIER.R.A-@smtpgty.bwi.wec.com (LANIER.R.A-)
To: hfsig@tapr.org, ss@tapr.org
Subject: Sinewave generator source code
Content-Type: multipart/mixed; boundary="IMA.Boundary.273410538"

--IMA.Boundary.273410538
Content-Type: text/plain; charset=US-ASCII
Content-Transfer-Encoding: 7bit
Content-Description: cc:Mail note part

For what its worth, I found the article I was looking for (Radio Electronics, October 1991). The author used a C program to generate the data for the EPROM feeding into the DAC. His source code is shown below. I also attached a sample output file to this note. The EPROM is a standard 2716. Of course, the code can be modified to suit your needs.

I intend to use this to produce a LUT (Look Up Table) inside a Xilinx FPGA. I want to use the LUT as the basis for a QPSK modulator. If I can feed serial data into the Xilinx, I can then produce I and Q components and use these as inputs into a custom up/down counter. The output of the counter will feed into the LUT. The output of the LUT would then feed into an external DAC. I'm not sure if this can work, but I will anyone who cares about this now what results I came up with.

73s de
Tony, KE4ATO

```
/* This program calculates the value of the sine function
   offset so that the 4th and 1st quadrants cause a code
   from 0 to 255. Code is generated to fill a 2048 byte prom
   (2716 or equivalent) for a full circle of 2*pi radians.
   Other size memory may be used by changing the value of
   bytes in the declaration table.
*/

#include <stdio.h>
#include <math.h>

main ()
```

```
{
  double p=0;           /* phase input to sine function */
       double S=0;      /* output value of the true sine function */
       int s;           /* amplitude truncated to 8 bits */
       double sin();    /* true sine function */
       double pi=3.141592654;
       int addr=0;      /* address of EPROM */
       int bytes=2048;  /* size of EPROM in bytes */

       printf("          0    1    2    3    4    5    6    7    8");
       printf("    9    A    B    C    D    E    F  \n");
       while (addr < bytes)
         {
         if (addr % 16 == 0)
           printf("\n%4x   ", addr);
         p = 2.0*pi*( (double) addr)/( (double) bytes);
         S = 127.5*(1.0 + sin(p - pi/2.0));       /* gives 0 at -90 deg */
         s = ( (int) S);                          /* convert to integer */
         if (S - ( (double) s) >= 0.5)            /* rounds if necessary */
           s++;
         printf("   %2x", s);
         addr++;                                  /* increment address */
         }
      return (0);
      }
```

```
--IMA.Boundary.273410538
Content-Type: text/basic; name="output.txt"
Content-Transfer-Encoding: 7bit
Content-Description: MS-DOS text file
Content-Disposition: attachment; filename="output.txt"
```

	0	1	2	3	4	5	6	7	8	9	A	B	C	D	E	F
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
10	0	0	0	0	0	0	0	0	0	0	0	0	0	1	1	1
20	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
30	1	1	1	2	2	2	2	2	2	2	2	2	2	2	2	2
40	2	3	3	3	3	3	3	3	3	3	3	3	3	4	4	4
50	4	4	4	4	4	4	4	5	5	5	5	5	5	5	5	5
60	5	6	6	6	6	6	6	6	6	7	7	7	7	7	7	7
70	7	8	8	8	8	8	8	8	9	9	9	9	9	9	9	a
80	a	a	a	a	a	a	b	b	b	b	b	b	c	c	c	c
90	c	c	d	d	d	d	d	d	e	e	e	e	e	f	f	f
a0	f	f	f	10	10	10	10	10	11	11	11	11	11	12	12	12
b0	12	12	13	13	13	13	13	14	14	14	14	14	15	15	15	15
c0	15	16	16	16	16	17	17	17	17	17	18	18	18	18	19	19
d0	19	19	1a	1a	1a	1a	1b	1b	1b	1b	1b	1c	1c	1c	1c	1d
e0	1d	1d	1d	1e	1e	1e	1e	1f	1f	1f	1f	20	20	20	21	21
f0	21	21	22	22	22	22	23	23	23	23	24	24	24	25	25	25
100	25	26	26	26	26	27	27	27	28	28	28	28	29	29	29	2a
110	2a	2a	2a	2b	2b	2b	2c	2c	2c	2d	2d	2d	2d	2e	2e	2e
120	2f	2f	2f	30	30	30	30	31	31	31	32	32	32	33	33	33

130	34	34	34	34	35	35	35	36	36	36	37	37	37	38	38	38
140	39	39	39	3a	3a	3a	3b	3b	3b	3c	3c	3c	3d	3d	3d	3e
150	3e	3e	3f	3f	3f	40	40	40	41	41	41	42	42	42	43	43
160	43	44	44	44	45	45	45	46	46	47	47	47	48	48	48	49
170	49	49	4a	4a	4a	4b	4b	4b	4c	4c	4d	4d	4d	4e	4e	4e
180	4f	4f	4f	50	50	51	51	51	52	52	52	53	53	53	54	54
190	55	55	55	56	56	56	57	57	58	58	58	59	59	59	5a	5a
1a0	5a	5b	5b	5c	5c	5c	5d	5d	5d	5e	5e	5f	5f	5f	60	60
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1c0	67	67	67	68	68	69	69	69	6a	6a	6a	6b	6b	6c	6c	6c
1d0	6d	6d	6e	6e	6e	6f	6f	70	70	70	71	71	71	72	72	73
1e0	73	73	74	74	75	75	75	76	76	77	77	77	78	78	78	79
1f0	79	7a	7a	7a	7b	7b	7c	7c	7c	7d	7d	7e	7e	7e	7f	7f
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280	b0	b1	b1	b1	b2	b2	b2	b3	b3	b4	b4	b4	b5	b5	b5	b6
290	b6	b6	b7	b7	b7	b8	b8	b8	b9	b9	ba	ba	ba	bb	bb	bb
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2c0	c6	c7	c7	c7	c8	c8	c8	c9	c9	c9	ca	ca	ca	cb	cb	cb
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2e0	d0	d1	d1	d1	d2	d2	d2	d2	d3	d3	d3	d4	d4	d4	d5	d5
2f0	d5	d5	d6	d6	d6	d7	d7	d7	d7	d8	d8	d8	d9	d9	d9	d9
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320	e2	e2	e3	e3	e3	e3	e4	e4	e4	e4	e4	e5	e5	e5	e5	e6
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3b0	fb	fb	fb	fb	fc	fc	fc	fc	fc	fc	fc	fc	fc	fc	fc	fc
3c0	fd	fd	fd	fd	fd	fd	fd	fd	fd	fd	fd	fd	fd	fd	fe	fe
3d0	fe	fe	fe	fe	fe	fe	fe	fe	fe	fe	fe	fe	fe	fe	fe	fe
3e0	fe	fe	fe	fe	ff	ff	ff	ff	ff	ff	ff	ff	ff	ff	ff	ff
3f0	ff	ff	ff	ff	ff	ff	ff	ff	ff	ff	ff	ff	ff	ff	ff	ff
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4c0	ea	e9	e9	e9	e9	e8	e8	e8	e8	e8	e7	e7	e7	e7	e6	e6
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570	b6	b6	b5	b5	b5	b4	b4	b4	b3	b3	b2	b2	b2	b1	b1	b1
580	b0	b0	b0	af	af	ae	ae	ae	ad	ad	ad	ac	ac	ac	ab	ab
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5d0	92	92	91	91	91	90	90	8f	8f	8f	8e	8e	8e	8d	8d	8c
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650	61	60	60	5f	5f	5f	5e	5e	5d	5d	5d	5c	5c	5c	5b	5b
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670	55	54	54	53	53	53	52	52	52	51	51	51	50	50	4f	4f
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6a0	43	43	43	42	42	42	41	41	41	40	40	40	3f	3f	3f	3e
6b0	3e	3e	3d	3d	3d	3c	3c	3c	3b	3b	3b	3a	3a	3a	39	39
6c0	39	38	38	38	37	37	37	36	36	36	35	35	35	34	34	34
6d0	34	33	33	33	32	32	32	31	31	31	30	30	30	30	2f	2f
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710	21	21	21	20	20	20	1f	1f	1f	1f	1e	1e	1e	1e	1d	1d
720	1d	1d	1c	1c	1c	1c	1b	1b	1b	1b	1b	1a	1a	1a	1a	19
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740	15	15	15	15	15	14	14	14	14	14	13	13	13	13	13	12
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780	a	a	9	9	9	9	9	9	9	8	8	8	8	8	8	8
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7b0	4	4	4	4	3	3	3	3	3	3	3	3	3	3	3	3
7c0	2	2	2	2	2	2	2	2	2	2	2	2	2	2	1	1

7d0	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
7e0	1	1	1	1	0	0	0	0	0	0	0	0	0	0	0	0
7f0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0

--IMA.Boundary.273410538--

From forrerj@ucs.orst.edu Mon Jun 17 10:40:27 1996
Received: from ucs.orst.edu (forrerj@UCS.ORST.EDU [128.193.4.5]) by tapr.org
(8.7.5/8.7.3/1.9) with SMTP id KAA17669 for <hfsig@tapr.org>; Mon, 17 Jun 1996
10:40:23 -0500 (CDT)
Received: by ucs.orst.edu; (5.65v3.2/1.1.8.2/13Mar96-1233PM)
id AA28166; Mon, 17 Jun 1996 08:40:14 -0700
Date: Mon, 17 Jun 1996 08:40:14 -0700 (PDT)
From: Johan Forrer <forrerj@ucs.orst.edu>
To: hfsig@tapr.org
Subject: Re: [HFSIG:1218] BTL performance tests
In-Reply-To: <H000029201ddaff7@MHS>
Message-Id: <Pine.OSF.3.91.960617083834.20926A-100000@ucs.orst.edu>
Mime-Version: 1.0
Content-Type: TEXT/PLAIN; charset=US-ASCII

Hi Robert,

Congratulations! - Nice writeup.

Is the program generally available yet, or what are your future plans.

Keep up the good work.

--Johan

On Mon, 17 Jun 1996 Robert.Glassey@nmp.nokia.com wrote:

```
> Hi all,
>
> Over the weekend I did a few performance tests on my PC-
> Sound Blaster based DSP RTTY modem (BTL), and compared it
> with my PK232MBX. Here's the results. Any coments welcome.
>
>                                  PK232MBX     BTL Ver 0.2
>
> 1  Noise performance:             15dB           12dB
>    Eb/No for BER=10^-2
>
> 2  Adjacent channel rejection (+500Hz 98 baud RTTY)
>
> 2a For 3dB rise in 10^-2 Eb/No:
>    Square pulse interferer         9dB           30dB
>    Raised cosine pulse interferer   -            32dB
>
> 2b For clean wanted signal, 10^-2 BER:
>    Square pulse interferer        16dB           32dB
```

```
>    Raised cosine pulse interferer  -            35dB
>
>
> Observations:
>
> 1. The BER taken includes errors due to both sync
>    errors and data bit errors. For both modems
>    relative contributions were about 50-50.
>    If a bit synchronous mode were used rather than
>    asynchronous baudot, BER would be half. My
>    measurements show this is equivelent to a 1dB
>    improvement in Eb/No for both modems.
>
> 2. Two different interferer types were used. Both were
>    phase continous FSK, 98 baud baudot, centred 500Hz
>    above the wanted signal (wanted 1275/1445, interferer
>    1775/1945) Standard RTTY has a square pulse shape
>    resulting in  high adjacent channel power. (-30dB @
>    500Hz) This limited measurements (and is a real on air
>    limit too) However I tried shaping the interferering
>    FSK pulses with raised cosine edges (shaped 64 of 82
>    samples). ACP was reduced to -45dB @ 500Hz. Using this
>    interferer, the true performance of the modem could be
>    determined. This shaping reduced the interference RMS
>    level to 72% and this was taken into account.
>
>    The minimum signal level without intefernce was found
>    to be 1 ADC step peak to peak, with dithering from the
>    noise of almost 3 ADC steps peak-peak. (signals were
>    calculated in 16 bits then quantised to 8 bits to allow
>    noise dithering). At this level performance was still
>    12dB Eb/No @ BER 10^-2. This minimum signal level was
>    44dB below the inteference level used above. Thus
>    quantisation was not a limiting factor in the above
>    measurements. Lower levels were possible but Eb/No was
>    worsened and quantisation boundries became significant.
>
>
> Measurement Technique:
>
> 1.  Eb/No measurements
>
> Sampling rate was 8KHz
>
> The signal was 45 baud Baudot, phase continuous AFSK, 1275Hz
> space and 1445Hz mark (170Hz shift). RMS to peak ratio =
> 0.71
>
> The noise was generated by a 24bit maximum length PRN
> generator (poly=0x8D0000), doing 8 shifts per sample to get
> an 8 bit uniform random noise source. RMS to peak ratio of a
> uniform random noise source = 0.58
>
```

```
> I have assumed a uniform distribution is OK since after
> bandpass filtering and integration the noise distribution
> aproaches gausian. (roughly speaking, the noise is averaged
> over the bit period, ie 178 samples, making the noise
> gausian by summation)
>
> For Eb/No calculations I have used the RMS levels at the
> 8kHz sample rate.
>
> Peak-peak levels of signal=45 and noise=128 becomes 12db
> Eb/No thus:
>
> Eb/No =  20*log10 ((45/2 * 0.71) / (128/2 * 0.58))
>          + 10*log10(4000 Hz) - 10*log10(45 BPS)
>       =  12.2 dB
>
>
> 2. Adjacent channel measurements
>
>    The signals are described above.
>
>    The interferer baud rate of 98 baud was chosen to try to
>    ovoid synchronisation between the wanted and interferer
>    so errors would appear more random, hopefully giving a
>    more accurate BER. However it does make the interferer
>    wider, but pulse shaping dealt with that. This may not
>    be the best method.
>
>    Results are given for two different cases. 1. For a
>    3dB rise in the Eb/No required for 10^-2 BER, and
>    2. The interference level that gives a BER of 10^-2
>    when the wanted signal is clean. These figures show
>    respectivly the degradation in weak signal performance,
>    and the maximum tolerable adjacent channel signal level
>    when the wanted signal is clean.
>
>
>
> So, 12dB Eb/No doesn't sound all that hot, does it? But is this what
> should be expected for FSK without ECC? Eb/No would be 11dB if a bit
> synchronous mode was used. These are for a BER of 10^-2.
>
> Cheers,
>
> Rob
>
>
>
>
>
>

From forrerj@ucs.orst.edu Mon Jun 17 11:09:42 1996
```

Received: from ucs.orst.edu (forrerj@UCS.ORST.EDU [128.193.4.5]) by tapr.org (8.7.5/8.7.3/1.9) with SMTP id LAA19201 for <hfsig@tapr.org>; Mon, 17 Jun 1996 11:09:39 -0500 (CDT)
Received: by ucs.orst.edu; (5.65v3.2/1.1.8.2/13Mar96-1233PM)
id AA06172; Mon, 17 Jun 1996 09:09:35 -0700
Date: Mon, 17 Jun 1996 09:09:34 -0700 (PDT)
From: Johan Forrer <forrerj@ucs.orst.edu>
To: hfsig@tapr.org
Subject: Re: [HFSIG:1219] Re: Further thoughts on pulse shaping for HF
In-Reply-To: <199606171016.DAA00188@unix.ka9q.ampr.org>
Message-Id: <Pine.OSF.3.91.960617084032.20926B-100000@ucs.orst.edu>
Mime-Version: 1.0
Content-Type: TEXT/PLAIN; charset=US-ASCII

Hi Phil,

On Mon, 17 Jun 1996, Phil Karn wrote:

> FYI, more about offset QPSK vs conventional QPSK.
>
> It looks as though the advantages of staggered (offset) QPSK are
> outweighted by the disadvantages, so I've switched to straight QPSK.
>
> There are two main advantages of offset QPSK: 1) less envelope droop
> when the signal is bandlimited, which turns into less sideband
> regrowth when the signal is nonlinearly amplified, and 2) less
> crosstalk between channels when the carrier phase reference is
> inaccurate.
>
> I originally chose SQPSK for reason #2; I don't really care too much
> about #1 since I am going to operate in a power-controlled regime
> where I assume there'll be a fair amount of transmit headroom most of
> the time, hence plenty of linearity. And even if there were sideband
> regrowth it wouldn't be as important a source of interference since
> I'm already operating at very low S/N ratios thanks to power control
> and coding.
>
> But it turns out that the crosstalk resistance of SQPSK is largely
> mitigated by greatly increased pattern-sensitive jitter in the carrier
> recovery system. In other words, although SQPSK has increased carrier
> phase jitter tolerance, it inherently increases the jitter of the
> recovered carrier which squanders the improved performance.
>
> I couldn't find a precise quantitative formulation for this, but from
> what I saw it looks like any gains SQPSK might have (especially at the
> very low Es/N0s I'm using) are small or even negative. And the last
> straw was that SQPSK is harder to implement than straight QPSK. So
> I've switched to straight QPSK.
>
> Phil
>
>

>
>

Your initial considerations for using the offset format seems quite valid
if one perhaps can overcome the side effects you mention.

About QPSK: The clean compact signal format that I see here is very
encouraging. Using raised cosine pulse shaping is a bit of a hassle, but
well worth it. I think if one can address the issue of non-linear
amplification, that this may not be a bad way to go.

One other thing that I experimented with this weekend was
non-coherent detection of DQPSK. After getting it to work, I was quite
surprized at how well it worked out. The constellations were tight
and text-book style. There is the expected recovery loss (i.e. lower
dynamic range) by giving up carrier extraction, however, I am of the
opinion that in the end it may work out best for HF especially
when used in conjunction with good coding.

The issue about the use of an unmodulated carrier for doppler compensation:
One of the first things that I do in my modem is to create an analytic
signal, with a frequency shifter stage built into the front end. The
frequency shifter is controlled by a doppler extractor algorithm.
Doppler estimation is modelled as a second order AR process derived from
the special unmodulated carrier after filtering by a narrow-band filter.
The frequency estimate is obtained solving two simulataneous equations
(Yule-Walker). There are provisions to deal with times when the carrier has
faded - pretty much a very heavy damped system similar to what I do with
clock extraction.

Good luck with your project.

--Johan

From N0AOT@aol.com Mon Jun 17 13:03:12 1996
Received: from emout09.mail.aol.com (emout09.mx.aol.com [198.81.11.24]) by
tapr.org (8.7.5/8.7.3/1.9) with SMTP id NAA24196 for <hfsig@tapr.org>; Mon, 17 Jun
1996 13:03:07 -0500 (CDT)
From: N0AOT@aol.com
Received: by emout09.mail.aol.com (8.6.12/8.6.12) id OAA10451; Mon, 17 Jun 1996
14:02:57 -0400
Date: Mon, 17 Jun 1996 14:02:57 -0400
Message-ID: <960617140256_558017278@emout09.mail.aol.com>
To: hfsig@tapr.org, Robert.Glassey@nmp.nokia.com
Subject: Re: [HFSIG:1218] BTL performance tests

Hi Bob--

In a message dated 96-06-17 06:48:58 EDT, you write:

>Hi all,
>
>Over the weekend I did a few performance tests on my PC-

>Sound Blaster based DSP RTTY modem (BTL), and compared it
>with my PK232MBX. Here's the results. Any coments welcome.
>
>

Very interesting weekend project you had there! I am not familiar with the BTL software you mentioned. Who is the author, and where is it available? Can the hardware interupt and io address of the sound card be changed so that it can be run on a notebook computer that has a "Sound Blaster compatible" chip?

I have a Motorola DSP56002EVM board, and would like to run similar tests...

--73's de Bob Carlson, n0aot@amsat.org

From LANIER.R.A-@smtpgty.bwi.wec.com Mon Jun 17 14:06:24 1996
Received: from tron.bwi.wec.com (tron.bwi.wec.com [129.228.4.1]) by tapr.org
(8.7.5/8.7.3/1.9) with SMTP id OAA26860 for <hfsig@tapr.org>; Mon, 17 Jun 1996
14:06:12 -0500 (CDT)
Received: from smtpgty.bwi.wec.com by tron.bwi.wec.com;
(5.65/1.1.8.2/31May95-0229PM)
     id AA31591; Mon, 17 Jun 1996 14:20:15 -0400
Received: from ccMail by smtpgty.bwi.wec.com
  (IMA Internet Exchange 2.0 Enterprise) id 1C5AC3F0; Mon, 17 Jun 96 15:04:31
-0400
Mime-Version: 1.0
Date: Mon, 17 Jun 1996 14:56:48 -0400
Message-Id: <1C5AC3F0.1858@smtpgty.bwi.wec.com>
From: LANIER.R.A-@smtpgty.bwi.wec.com (LANIER.R.A-)
Subject: Re: [HFSIG:1222] Re: BTL performance tests
To: hfsig@tapr.org
Content-Type: text/plain; charset=US-ASCII
Content-Transfer-Encoding: 7bit
Content-Description: cc:Mail note part

Johan,

I think you have me confused with someone else. I didn't write the text below - wish I did though. Its pretty impressive!

73s de
Tony, KE4ATO

P.S. My first name is Robert, but I go by Tony (middle name Anthony) in all of my correspondence.

______________________________ Reply Separator _________________________________
Subject: [HFSIG:1222] Re: BTL performance tests
Author: hfsig@tapr.org at BALT.SMTP
Date: 6/17/96 10:49 AM

Hi Robert,

Congratulations! - Nice writeup.

Is the program generally available yet, or what are your future plans.

Keep up the good work.

--Johan

On Mon, 17 Jun 1996 Robert.Glassey@nmp.nokia.com wrote:

```
> Hi all,
>
> Over the weekend I did a few performance tests on my PC-
> Sound Blaster based DSP RTTY modem (BTL), and compared it
> with my PK232MBX. Here's the results. Any coments welcome.
>
>                                  PK232MBX     BTL Ver 0.2
>
> 1  Noise performance:             15dB           12dB
>    Eb/No for BER=10^-2
>
> 2  Adjacent channel rejection (+500Hz 98 baud RTTY)
>
> 2a For 3dB rise in 10^-2 Eb/No:
>    Square pulse interferer         9dB           30dB
>    Raised cosine pulse interferer   -            32dB
>
> 2b For clean wanted signal, 10^-2 BER:
>    Square pulse interferer        16dB           32dB
>    Raised cosine pulse interferer  -             35dB
>
>
> Observations:
>
> 1. The BER taken includes errors due to both sync
>    errors and data bit errors. For both modems
>    relative contributions were about 50-50.
>    If a bit synchronous mode were used rather than
>    asynchronous baudot, BER would be half. My
>    measurements show this is equivelent to a 1dB
>    improvement in Eb/No for both modems.
>
> 2. Two different interferer types were used. Both were
>    phase continous FSK, 98 baud baudot, centred 500Hz
>    above the wanted signal (wanted 1275/1445, interferer
>    1775/1945) Standard RTTY has a square pulse shape
>    resulting in  high adjacent channel power. (-30dB @
>    500Hz) This limited measurements (and is a real on air
>    limit too) However I tried shaping the interferering
```

```
>    FSK pulses with raised cosine edges (shaped 64 of 82
>    samples). ACP was reduced to -45dB @ 500Hz. Using this
>    interferer, the true performance of the modem could be
>    determined. This shaping reduced the interference RMS
>    level to 72% and this was taken into account.
>
>    The minimum signal level without intefernce was found
>    to be 1 ADC step peak to peak, with dithering from the
>    noise of almost 3 ADC steps peak-peak. (signals were
>    calculated in 16 bits then quantised to 8 bits to allow
>    noise dithering). At this level performance was still
>    12dB Eb/No @ BER 10^-2. This minimum signal level was
>    44dB below the inteference level used above. Thus
>    quantisation was not a limiting factor in the above
>    measurements. Lower levels were possible but Eb/No was
>    worsened and quantisation boundries became significant.
>
>
> Measurement Technique:
>
> 1.  Eb/No measurements
>
> Sampling rate was 8KHz
>
> The signal was 45 baud Baudot, phase continuous AFSK, 1275Hz
> space and 1445Hz mark (170Hz shift). RMS to peak ratio =
> 0.71
>
> The noise was generated by a 24bit maximum length PRN
> generator (poly=0x8D0000), doing 8 shifts per sample to get
> an 8 bit uniform random noise source. RMS to peak ratio of a
> uniform random noise source = 0.58
>
> I have assumed a uniform distribution is OK since after
> bandpass filtering and integration the noise distribution
> aproaches gausian. (roughly speaking, the noise is averaged
> over the bit period, ie 178 samples, making the noise
> gausian by summation)
>
> For Eb/No calculations I have used the RMS levels at the
> 8kHz sample rate.
>
> Peak-peak levels of signal=45 and noise=128 becomes 12db
> Eb/No thus:
>
> Eb/No =  20*log10 ((45/2 * 0.71) / (128/2 * 0.58))
>          + 10*log10(4000 Hz) - 10*log10(45 BPS)
>       =  12.2 dB
>
>
> 2. Adjacent channel measurements
>
>    The signals are described above.
```

>
> The interferer baud rate of 98 baud was chosen to try to
> ovoid synchronisation between the wanted and interferer
> so errors would appear more random, hopefully giving a
> more accurate BER. However it does make the interferer
> wider, but pulse shaping dealt with that. This may not
> be the best method.
>
> Results are given for two different cases. 1. For a
> 3dB rise in the Eb/No required for 10^-2 BER, and
> 2. The interference level that gives a BER of 10^-2
> when the wanted signal is clean. These figures show
> respectivly the degradation in weak signal performance,
> and the maximum tolerable adjacent channel signal level
> when the wanted signal is clean.
>
>
>
> So, 12dB Eb/No doesn't sound all that hot, does it? But is this what
> should be expected for FSK without ECC? Eb/No would be 11dB if a bit
> synchronous mode was used. These are for a BER of 10^-2.
>
> Cheers,
>
> Rob
>
>
>
>
>
>

From LANIER.R.A-@smtpgty.bwi.wec.com Mon Jun 17 16:22:47 1996
Received: from tron.bwi.wec.com (tron.bwi.wec.com [129.228.4.1]) by tapr.org
(8.7.5/8.7.3/1.9) with SMTP id QAA03634 for <hfsig@tapr.org>; Mon, 17 Jun 1996
16:21:57 -0500 (CDT)
Received: from smtpgty.bwi.wec.com by tron.bwi.wec.com;
(5.65/1.1.8.2/31May95-0229PM)
id AA23940; Mon, 17 Jun 1996 16:36:09 -0400
Received: from ccMail by smtpgty.bwi.wec.com
(IMA Internet Exchange 2.0 Enterprise) id 1C5CC660; Mon, 17 Jun 96 17:21:42
-0400
Mime-Version: 1.0
Date: Mon, 17 Jun 1996 17:19:52 -0400
Message-Id: <1C5CC660.1858@smtpgty.bwi.wec.com>
From: LANIER.R.A-@smtpgty.bwi.wec.com (LANIER.R.A-)
Subject: Re: [HFSIG:1224] Re: BTL performance tests
To: hfsig@tapr.org
Content-Type: text/plain; charset=US-ASCII
Content-Transfer-Encoding: 7bit
Content-Description: cc:Mail note part

I am not the author of this software project! I don't know how my name got mixed up.

Tony, KE4ATO

______________________________ Reply Separator _________________________________
Subject: [HFSIG:1224] Re: BTL performance tests
Author: hfsig@tapr.org at BALT.SMTP
Date: 6/17/96 1:15 PM

Hi Bob--

In a message dated 96-06-17 06:48:58 EDT, you write:

>Hi all,
>
>Over the weekend I did a few performance tests on my PC-
>Sound Blaster based DSP RTTY modem (BTL), and compared it
>with my PK232MBX. Here's the results. Any coments welcome.
>
>

Very interesting weekend project you had there! I am not familiar with the BTL software you mentioned. Who is the author, and where is it available? Can the hardware interupt and io address of the sound card be changed so that it can be run on a notebook computer that has a "Sound Blaster compatible" chip?

I have a Motorola DSP56002EVM board, and would like to run similar tests...

--73's de Bob Carlson, n0aot@amsat.org

From karn@qualcomm.com Mon Jun 17 22:37:52 1996
Received: from servo.qualcomm.com (servo.qualcomm.com [129.46.128.14]) by tapr.org (8.7.5/8.7.3/1.9) with ESMTP id WAA17713 for <hfsig@tapr.org>; Mon, 17 Jun 1996 22:37:43 -0500 (CDT)
Received: (from karn@localhost) by servo.qualcomm.com (8.7.5/1.0/8.7.2/1.9) id UAA02588; Mon, 17 Jun 1996 20:37:10 -0700 (PDT)
Date: Mon, 17 Jun 1996 20:37:10 -0700 (PDT)
From: Phil Karn <karn@qualcomm.com>
Message-Id: <199606180337.UAA02588@servo.qualcomm.com>
To: hfsig@tapr.org
In-reply-to: <Pine.OSF.3.91.960617084032.20926B-100000@ucs.orst.edu> (message from Johan Forrer on Mon, 17 Jun 1996 11:22:58 -0500 (CDT))

Subject: Re: [HFSIG:1223] Re: Further thoughts on pulse shaping for HF

Johan,

Noncoherent schemes have a lot going for them on fading, dispersive

channels. Not only because it's hard to extract a carrier phase that's rapidly changing, but also because the dynamic range of the fading tends to swamp the relatively small Eb/N0 ratio differences between coherent and noncoherent modulation that are otherwise important on severely power-constrained nonfading AWGN channels.

The three most important things on a fading channel are diversity, diversity and diversity. :-) Diversity can be in time, space or frequency. Coding helps promotes diversity, particularly in time by spreading out the influence of one data bit over many symbols (only some of which may get through). And even QPSK can help by simply creating more slots for those coded symbols in the first place.

Phil

From karn@qualcomm.com Mon Jun 17 22:58:10 1996
Received: from servo.qualcomm.com (servo.qualcomm.com [129.46.128.14]) by tapr.org (8.7.5/8.7.3/1.9) with ESMTP id WAA18215 for <hfsig@tapr.org>; Mon, 17 Jun 1996 22:58:07 -0500 (CDT)
Received: (from karn@localhost) by servo.qualcomm.com (8.7.5/1.0/8.7.2/1.9) id UAA02630; Mon, 17 Jun 1996 20:57:35 -0700 (PDT)
Date: Mon, 17 Jun 1996 20:57:35 -0700 (PDT)
From: Phil Karn <karn@qualcomm.com>
Message-Id: <199606180357.UAA02630@servo.qualcomm.com>
To: hfsig@tapr.org
In-reply-to: <Pine.OSF.3.91.960617084032.20926B-100000@ucs.orst.edu> (message from Johan Forrer on Mon, 17 Jun 1996 11:22:58 -0500 (CDT))
Subject: Re: [HFSIG:1223] Re: Further thoughts on pulse shaping for HF

>One of the first things that I do in my modem is to create an analytic
>signal, with a frequency shifter stage built into the front end. The
>frequency shifter is controlled by a doppler extractor algorithm.

This is exactly my approach also. The frequency shifter is at present set to a fixed value based on the estimated frequency of the carrier burst in the packet preamble. This assumes that any residual error is small compared to the data rate (it is), so it can be tracked out after filtering and downsampling from 8x to 1x the symbol rate. (The downsampling starts once I have obtained one-shot symbol sync from the preamble).

Although all further processing has to be done with complex arithmetic, performance-wise I think I'm still ahead after the downsampling.

Eventually I will probably have to add a doppler chirper to the frequency shifter driven by an orbit tracking routine. This is relatively straightforward given accurate orbital elements, time and station location.

Phil

From forrerj@ucs.orst.edu Mon Jun 17 23:54:01 1996
Received: from ucs.orst.edu (root@UCS.ORST.EDU [128.193.4.5]) by tapr.org
(8.7.5/8.7.3/1.9) with SMTP id XAA21367 for <hfsig@tapr.org>; Mon, 17 Jun 1996
23:53:53 -0500 (CDT)
Received: from p08.t0.monrotel.com by ucs.orst.edu;
(5.65v3.2/1.1.8.2/13Mar96-1233PM)
     id AA18775; Mon, 17 Jun 1996 21:53:44 -0700
Message-Id: <1.5.4.16.19960618055458.4c9f67fc@ucs.orst.edu>
X-Sender: forrerj@ucs.orst.edu
X-Mailer: Windows Eudora Light Version 1.5.4 (16)
Mime-Version: 1.0
Content-Type: text/plain; charset="us-ascii"
Date: Mon, 17 Jun 1996 21:54:58 -0800
To: hfsig@tapr.org
From: Johan Forrer <forrerj@ucs.orst.edu>
Subject: Re: [HFSIG:1225] Re: BTL performance tests

At 02:21 PM 6/17/96 -0500, you wrote:
>   Johan,
>
>   I think you have me confused with someone else. I didn't write the
>   text below - wish I did though. Its pretty impressive!
>
>   73s de
>   Tony, KE4ATO
>
>   P.S. My first name is Robert, but I go by Tony (middle name Anthony)
>   in all of my correspondence.
>
>

Strange things do happen :-)

My reply was directed at Robert (Rob) Glassey's fine work (please see below)
- somehow we got our lines crossed - no harm done. Still a nice piece of
work Rob.

--Johan

>Hi Robert,
>
>Congratulations! - Nice writeup.
>
>Is the program generally available yet, or what are your future plans.
>
>Keep up the good work.
>

...... some lines deleted

>
>

>On Mon, 17 Jun 1996 Robert.Glassey@nmp.nokia.com wrote:
>

...... some more lines deleted

From Robert.Glassey@nmp.nokia.com Tue Jun 18 04:22:31 1996
Received: from noknic.nokia.com (noknic.nokia.com [131.228.6.10]) by tapr.org
(8.7.5/8.7.3/1.9) with SMTP id EAA03763 for <hfsig@tapr.org>; Tue, 18 Jun 1996
04:22:27 -0500 (CDT)
From: Robert.Glassey@nmp.nokia.com
Received: from samail01.nmp.nokia.com (samail01.nmp.nokia.com [131.228.240.6]) by
noknic.nokia.com (8.6.9/8.6.9) with ESMTP id MAA27649 for <hfsig@tapr.org>; Tue,
18 Jun 1996 12:21:49 +0300
Received: from by samail01.nmp.nokia.com with SMTP
     (1.37.109.16/16.2) id AA262409595; Tue, 18 Jun 1996 12:19:55 +0300
X-Openmail-Hops: 2
Date: Tue, 18 Jun 96 10:15:50 +0100
Message-Id: <H000029201dfc18f@MHS>
In-Reply-To: <Pine.OSF.3.91.960617083834.20926A-100000@ucs.orst.edu>
Subject: [HFSIG:1222] Re: BTL performance tests
Sender: Robert.Glassey@nmp.nokia.com
To: hfsig@tapr.org

> Hi Robert,
>
> Congratulations! - Nice writeup.
>
> Is the program generally available yet, or what are your future plans.
>
> Keep up the good work.
>
> --Johan
>
>
> On Mon, 17 Jun 1996 Robert.Glassey@nmp.nokia.com wrote:
>
> > Hi all,
> >
> > Over the weekend I did a few performance tests on my PC-
> > Sound Blaster based DSP RTTY modem (BTL), and compared it
> > with my PK232MBX. Here's the results. Any coments welcome.

Thanks, Its still on beta test, with V0.2beta out very soon (I hope)
- Sorry to those testers still waiting for the next version,
unfortunatly my weekend projects don't always get the time they deserve.

The release version 1.0, should be ready in a copule of weeks. This
version recieves standard baudot only, with variable baud, shift and
centre frequency, plus a few other features.

My next release (could be 6 months away) will have TX, and a wide shift
mode for comercial news services etc. I hope to include AMTOR FEC and
ARQ modes as well. Plus any improvements in performance I can find,

probably a bit-synchronous demod mode (for standard 7.5 bit baudot), and
maybe an improved tuning indicator, auto tuning, and mode/baud
detection.

This was my intention from the start for a first release, but things
have taken a while, and I think people want it as it is right now.

Ultimatly I would like to use BTL as a platform for more complex modes
such as PACTOR II or Clover II or other multicarrier modes that may be
developed through HFSIG. I'm also keen to implement very low SNR modes,
such as DBPSK with heaps of ECC using a type II hybrid ARQ protocol.

I intend to release up to version 2 as public domain freeware.

Cheers,

Rob Glassey, G0VTQ

From BRYD@KIDD.CO.ZA Tue Jun 18 04:26:46 1996
Received: from igw01 (igw01.kidd.co.za [192.96.246.1]) by tapr.org
(8.7.5/8.7.3/1.9) with SMTP id EAA04022 for <hfsig@tapr.org>; Tue, 18 Jun 1996
04:26:41 -0500 (CDT)
Received: from KIDD.CO.ZA by igw01 with smtp
     (Smail3.1.29.1 #3) id m0uVx3H-000PAWC; Tue, 18 Jun 96 11:26 GMT+0200
Received: from KenMail-Message_Server by KIDD.CO.ZA
     with Novell_GroupWise; Tue, 18 Jun 1996 11:33:08 +0200
Message-Id: <s1c693f3.043@KIDD.CO.ZA>
X-Mailer: Novell GroupWise 4.1
Date: Tue, 18 Jun 1996 11:26:45 +0200
From: Danie Brynard <BRYD@KIDD.CO.ZA>
To: hfsig@tapr.org
Subject: Well, the standard Intel programmer's references are pretty

          good. They tell you all you really need

Well, the standard Intel programmer's references are pretty good. They tell you
all you really need to know to
optimize DSP code on the
Pentium, such as instruction clock counts and considerations on how to keep the
pipelines full.

Phil

I see. Thanks. 73 danie

From Robert.Glassey@nmp.nokia.com Tue Jun 18 05:38:14 1996
Received: from noknic.nokia.com (noknic.nokia.com [131.228.6.10]) by tapr.org
(8.7.5/8.7.3/1.9) with SMTP id FAA05769 for <hfsig@tapr.org>; Tue, 18 Jun 1996
05:38:11 -0500 (CDT)
From: Robert.Glassey@nmp.nokia.com
Received: from samail01.nmp.nokia.com (samail01.nmp.nokia.com [131.228.240.6]) by
noknic.nokia.com (8.6.9/8.6.9) with ESMTP id MAA29755 for <hfsig@tapr.org>; Tue,

18 Jun 1996 12:52:02 +0300
Received: from by samail01.nmp.nokia.com with SMTP
     (1.37.109.16/16.2) id AA298861408; Tue, 18 Jun 1996 12:50:08 +0300
X-Openmail-Hops: 2
Date: Tue, 18 Jun 96 10:45:45 +0100
Message-Id: <H000029201dfdd45@MHS>
In-Reply-To: <960617140256_558017278@emout09.mail.aol.com>
Subject: [HFSIG:1224] Re: BTL performance tests
Sender: Robert.Glassey@nmp.nokia.com
To: hfsig@tapr.org

> Hi Bob--
>
> In a message dated 96-06-17 06:48:58 EDT, you write:
>
> >Hi all,
> >
> >Over the weekend I did a few performance tests on my PC-
> >Sound Blaster based DSP RTTY modem (BTL), and compared it
> >with my PK232MBX. Here's the results. Any coments welcome.
> >
> >
>
> Very interesting weekend project you had there! I am not familiar
> with the BTL software you mentioned. Who is the author, and where is
> it available? Can the hardware interupt and io address of the sound
> card be changed so that it can be run on a notebook computer that has
> a "Sound Blaster compatible" chip?
>
> I have a Motorola DSP56002EVM board, and would like to run similar
> tests...
>
> --73's de Bob Carlson, n0aot@amsat.org

Hi Bob,

BTL is not on general release yet, but soon will be. BTL is my 'baby'
and has taken me since around August last year to write (very slowly
since its a weekend project). When released, I hope to get it onto an
FTP site somewhere, maybe the TAPR site? This version is FREE!

Soundblaster parameters are user configurable, and it accepts the
standard sound blaster enviroment variable BLASTER=....

It should work on any soundblaster compatible from the original v1.0 to
the latest AWE32.

It should run on your notebook if its fast enough. To date, the slowest
machine I've tried is a 486sx25 and that works fine. If speed is a
problem I may release a cut down 1.5 version which should run faster.

The tests were done with a C program that generated test signals plus
noise that I used as my signal generator for the tests.

The test signal was played on the SB for the PK 232, and taken directly
from disk, in real time, for the BTL tests, by substituting the
soundblaster DMA for a double buffered disk routine.

I also have a DAT recorder that I have used in the past for testing.
I'll have to repeat these BTL tests using recorded signals.

Cheers,

Rob, G0VTQ

From BRYD@KIDD.CO.ZA Wed Jun 19 08:46:09 1996
Received: from igw01 (igw01.kidd.co.za [192.96.246.1]) by tapr.org
(8.7.5/8.7.3/1.9) with SMTP id IAA09757 for <hfsig@tapr.org>; Wed, 19 Jun 1996
08:46:04 -0500 (CDT)
Received: from KIDD.CO.ZA by igw01 with smtp
     (Smail3.1.29.1 #3) id m0uWNsX-000PHoC; Wed, 19 Jun 96 16:05 GMT+0200
Received: from KenMail-Message_Server by KIDD.CO.ZA
     with Novell_GroupWise; Wed, 19 Jun 1996 15:52:34 +0200
Message-Id: <s1c82242.070@KIDD.CO.ZA>
X-Mailer: Novell GroupWise 4.1
Date: Wed, 19 Jun 1996 15:46:16 +0200
From: Danie Brynard <BRYD@KIDD.CO.ZA>
To: hfsig@tapr.org
Subject:

I also have a DAT recorder that I have used in the past for testing.
I'll have to repeat these BTL tests using recorded signals.

Bob how well does the DAT recorder work and is it not too expensive ? I also have
a need for soemthing like that.
It is difficult with the sat signals as they pass very quickly...no time to
recompile and download hi :-)

danie zs6awk

From Robert.Glassey@nmp.nokia.com Wed Jun 19 10:34:23 1996
Received: from noknic.nokia.com (noknic.nokia.com [131.228.6.10]) by tapr.org
(8.7.5/8.7.3/1.9) with SMTP id KAA16025 for <hfsig@tapr.org>; Wed, 19 Jun 1996
10:34:15 -0500 (CDT)
From: Robert.Glassey@nmp.nokia.com
Received: from samail01.nmp.nokia.com (samail01.nmp.nokia.com [131.228.240.6]) by
noknic.nokia.com (8.6.9/8.6.9) with ESMTP id SAA26250 for <hfsig@tapr.org>; Wed,
19 Jun 1996 18:33:35 +0300
Received: from by samail01.nmp.nokia.com with SMTP
     (1.37.109.16/16.2) id AA104938299; Wed, 19 Jun 1996 18:31:39 +0300
X-Openmail-Hops: 2
Date: Wed, 19 Jun 96 16:30:26 +0100
Message-Id: <H000029201e2ac42@MHS>

In-Reply-To: <s1c82242.070@KIDD.CO.ZA>
Subject: [HFSIG:1233]
Sender: Robert.Glassey@nmp.nokia.com
To: hfsig@tapr.org

>> I also have a DAT recorder that I have used in the past for testing.
>> Bob how well does the DAT recorder work and is it not too expensive ?
> I also have a need for soemthing like that.
> It is difficult with the sat signals as they pass very quickly...no
> time to recompile and download hi :-)
>
> danie zs6awk

Hi Danie,

They work very well, even preserve timing on amtor/pactor. They are a little pricy, mine cost around $650 US, but it is a walkman model, (Sony TCD-D7). A hi-fi component style version should be cheaper. I find the walkman handy for taking a known test signal from one PC to another for testing on different sound cards etc. They are also usefull for recording a live QSO then replaying it over and over while modifying the program to test with a known signal. They are also available second hand occasionally, but beware of older models, reliability was a problem with them, and they're probably on their last legs now if they haven't already died.

If you have a sound blaster your best bet (and cheapest) would be to record the satilite on disk the play it back to the DSP. 10 minutes will take about 5MB. You may need make your DSP code more independent of the PC (buffer output etc) or get another PC!

Its actually quite simple to write a C program to play the sound file and act as a dumb terminal at the same time.

73's

Rob,

G0VTQ

From forrerj@ucs.orst.edu Wed Jun 19 11:17:09 1996
Received: from ucs.orst.edu (root@UCS.ORST.EDU [128.193.4.5]) by tapr.org (8.7.5/8.7.3/1.9) with SMTP id LAA17643 for <hfsig@tapr.org>; Wed, 19 Jun 1996 11:17:06 -0500 (CDT)
Received: from p00.t0.monrotel.com by ucs.orst.edu;
(5.65v3.2/1.1.8.2/13Mar96-1233PM)
 id AA19123; Wed, 19 Jun 1996 09:16:55 -0700
Message-Id: <1.5.4.16.19960619171812.3f8ffe56@ucs.orst.edu>
X-Sender: forrerj@ucs.orst.edu
X-Mailer: Windows Eudora Light Version 1.5.4 (16)
Mime-Version: 1.0
Content-Type: text/plain; charset="us-ascii"
Date: Wed, 19 Jun 1996 09:18:12 -0800

To: hfsig@tapr.org
From: Johan Forrer <forrerj@ucs.orst.edu>
Subject: Re: [HFSIG:1228] Re: Further thoughts on pulse shaping for HF

Phil,

......

>
>Although all further processing has to be done with complex
>arithmetic, performance-wise I think I'm still ahead after the
>downsampling.
>
>Eventually I will probably have to add a doppler chirper to the
>frequency shifter driven by an orbit tracking routine. This is
>relatively straightforward given accurate orbital elements, time and
>station location.

Sounds quite involved - fortunate that you have such a nice closed-form
predition method. I was wondering; are there any residual effects to deal
with due to spin motion of the satellite? some higher than doppler phasing
effects or does the circular polarized antennas take care of it?

Re: Doppler tracking on HF:
I'm looking into following reasonably fast-changing doppler effects, i.e., a
fraction of a symbol time. The method that I'm interested in for estimating
doppler comes from LPC methodology and the ideas are well developed -
extracting pitch, or in other words specifying a filter from the signal. I'm
not sure how well it will work for what I am doing, but I'm willing to give
it a chance.

Anyway, the modem will have to go on the back burner for a while - something
else has come up that needs to be done.

--Johan

From forrerj@ucs.orst.edu Wed Jun 19 11:17:09 1996
Received: from ucs.orst.edu (root@UCS.ORST.EDU [128.193.4.5]) by tapr.org
(8.7.5/8.7.3/1.9) with SMTP id LAA17644 for <hfsig@tapr.org>; Wed, 19 Jun 1996
11:17:06 -0500 (CDT)
Received: from p00.t0.monrotel.com by ucs.orst.edu;
(5.65v3.2/1.1.8.2/13Mar96-1233PM)
     id AA08169; Wed, 19 Jun 1996 09:17:00 -0700
Message-Id: <1.5.4.16.19960619171816.0d271ab0@ucs.orst.edu>
X-Sender: forrerj@ucs.orst.edu
X-Mailer: Windows Eudora Light Version 1.5.4 (16)
Mime-Version: 1.0
Content-Type: text/plain; charset="us-ascii"
Date: Wed, 19 Jun 1996 09:18:16 -0800
To: hfsig@tapr.org
From: Johan Forrer <forrerj@ucs.orst.edu>
Subject: Re: [HFSIG:1230] Re: BTL performance tests

Hi Rob,

>Ultimatly I would like to use BTL as a platform for more complex modes
>such as PACTOR II or Clover II or other multicarrier modes that may be
>developed through HFSIG. I'm also keen to implement very low SNR modes,
>such as DBPSK with heaps of ECC using a type II hybrid ARQ protocol.
>
>I intend to release up to version 2 as public domain freeware.
>
>
>Cheers,
>
>Rob Glassey, G0VTQ
>
>

Excellent!

Would you consider uploading a program version to HFSIG? If you do, also
please provide a ".txt" version for the folks that browse the files with
their web browser.

Keep us posted on your progress - sounds like good stuff.

--Johan

From karn@qualcomm.com Wed Jun 19 18:27:25 1996
Received: from servo.qualcomm.com (servo.qualcomm.com [129.46.128.14]) by tapr.org
(8.7.5/8.7.3/1.9) with ESMTP id SAA06837 for <hfsig@tapr.org>; Wed, 19 Jun 1996
18:27:22 -0500 (CDT)
Received: (from karn@localhost) by servo.qualcomm.com (8.7.5/1.0/8.7.2/1.9) id
QAA23790; Wed, 19 Jun 1996 16:26:49 -0700 (PDT)
Date: Wed, 19 Jun 1996 16:26:49 -0700 (PDT)
From: Phil Karn <karn@qualcomm.com>
Message-Id: <199606192326.QAA23790@servo.qualcomm.com>
To: hfsig@tapr.org
In-reply-to: <1.5.4.16.19960619171812.3f8ffe56@ucs.orst.edu> (message from Johan
Forrer on Wed, 19 Jun 1996 11:21:34 -0500 (CDT))
Subject: Re: [HFSIG:1235] Re: Further thoughts on pulse shaping for HF

>Sounds quite involved - fortunate that you have such a nice closed-form
>predition method. I was wondering; are there any residual effects to deal
>with due to spin motion of the satellite? some higher than doppler phasing
>effects or does the circular polarized antennas take care of it?

As a matter of fact, yes. James Miller has pointed out that the S-band
antenna on AO-13 is not mounted along the center of mass, so if the
antennas are not lined up on earth the antenna moves to and fro along
the line of sight as the spacecraft spins. This introduces a
sinusoidal doppler shift on the order of 5 Hz, which is significant to
me.

Of course, this may be somewhat academic as I probably won't get this
stuff on mode S before AO-13 burns up, and P3D is 3-axis stabilized.

Phil

From BRYD@KIDD.CO.ZA Thu Jun 20 06:29:12 1996
Received: from igw01 (igw01.kidd.co.za [192.96.246.1]) by tapr.org
(8.7.5/8.7.3/1.9) with SMTP id GAA11730 for <hfsig@tapr.org>; Thu, 20 Jun 1996
06:29:02 -0500 (CDT)
Received: from KIDD.CO.ZA by igw01 with smtp
    (Smail3.1.29.1 #3) id m0uWhv0-000PLoC; Thu, 20 Jun 96 13:28 GMT+0200
Received: from KenMail-Message_Server by KIDD.CO.ZA
    with Novell_GroupWise; Thu, 20 Jun 1996 13:35:46 +0200
Message-Id: <s1c953b2.040@KIDD.CO.ZA>
X-Mailer: Novell GroupWise 4.1
Date: Thu, 20 Jun 1996 13:28:28 +0200
From: Danie Brynard <BRYD@KIDD.CO.ZA>
To: hfsig@tapr.org
Subject:

If you have a sound blaster your best bet (and cheapest) would be to record the
satilite on disk the play it back to
the DSP. 10 minutes will take about 5MB. You may need make your DSP code more
independent of the
PC (buffer output etc) or get another PC!

Its actually quite simple to write a C program to play the sound file and act as a
dumb terminal at the same time.
=============================
I see. Yes I do have another PC. BTW what is the bandwidth of the DAT tape ? I
suppose it has now phase
distorsion ie wow & flutter ?

danie

From LANIER.R.A-@smtpgty.bwi.wec.com Thu Jun 20 10:41:54 1996
Received: from tron.bwi.wec.com (tron.bwi.wec.com [129.228.4.1]) by tapr.org
(8.7.5/8.7.3/1.9) with SMTP id KAA20149 for <hfsig@tapr.org>; Thu, 20 Jun 1996
10:41:49 -0500 (CDT)
Received: from smtpgty.bwi.wec.com by tron.bwi.wec.com;
(5.65/1.1.8.2/31May95-0229PM)
    id AA22177; Thu, 20 Jun 1996 11:27:48 -0400
Received: from ccMail by smtpgty.bwi.wec.com
  (IMA Internet Exchange 2.0 Enterprise) id 1C971760; Thu, 20 Jun 96 11:42:46
-0400
Mime-Version: 1.0
Date: Thu, 20 Jun 1996 11:37:50 -0400
Message-Id: <1C971760.1858@smtpgty.bwi.wec.com>
From: LANIER.R.A-@smtpgty.bwi.wec.com (LANIER.R.A-)
Subject: Re: [HFSIG:1238]

To: hfsig@tapr.org
Content-Type: text/plain; charset=US-ASCII
Content-Transfer-Encoding: 7bit
Content-Description: cc:Mail note part

I'm curious, what would this C program look like?

73s de
Tony, KE4ATO

______________________________ Reply Separator _________________________________
Subject: [HFSIG:1238]
Author: hfsig@tapr.org at BALT.SMTP
Date: 6/20/96 6:36 AM

If you have a sound blaster your best bet (and cheapest) would be to record the
satilite on disk the play it back to
the DSP. 10 minutes will take about 5MB. You may need make your DSP code more
independent of the
PC (buffer output etc) or get another PC!

Its actually quite simple to write a C program to play the sound file and act as
a dumb terminal at the same time.
=============================
I see. Yes I do have another PC. BTW what is the bandwidth of the DAT tape ? I
suppose it has now phase
distorsion ie wow & flutter ?

danie

From Robert.Glassey@nmp.nokia.com Thu Jun 20 12:40:16 1996
Received: from noknic.nokia.com (noknic.nokia.com [131.228.6.10]) by tapr.org
(8.7.5/8.7.3/1.9) with SMTP id MAA24162 for <hfsig@tapr.org>; Thu, 20 Jun 1996
12:40:12 -0500 (CDT)
From: Robert.Glassey@nmp.nokia.com
Received: from samail01.nmp.nokia.com (samail01.nmp.nokia.com [131.228.240.6]) by
noknic.nokia.com (8.6.9/8.6.9) with ESMTP id UAA10748; Thu, 20 Jun 1996 20:39:36
+0300
Received: from by samail01.nmp.nokia.com with SMTP
(1.37.109.16/16.2) id AA297632257; Thu, 20 Jun 1996 20:37:37 +0300
X-Openmail-Hops: 2
Date: Thu, 20 Jun 96 18:36:59 +0100
Message-Id: <H000029201e4bf8e@MHS>
In-Reply-To: <1C971760.1858@smtpgty.bwi.wec.com>
Subject: [HFSIG:1239] Re:
Sender: Robert.Glassey@nmp.nokia.com
To: hfsig@tapr.org, LANIER.R.A-@smtpgty.bwi.wec.com

>> Its actually quite simple to write a C program to play the sound file
>> and act as a dumb terminal at the same time.

> I'm curious, what would this C program look like?
>
> 73s de
> Tony, KE4ATO

There's libraries available on simtel for the SB stuff. It would look
something like this:

```
main()
{

soundblaster_reset(); /* also checks for card */

buffer_mem=faralloc(128*1024l); /* 64 k buffer when page aligned */

buffer=[page alignment of buffer_mem pointer ie to get XX00:0000]
/* this is so the DMA doesn't cross a 64K page boundry */

[setup interupt on SB IRQ]

[set DMA parameters - port commands]

[set sound blaster sample rate -port commands]

file=fopen("sat_rec","rb");

[read 32k from file into buffer]

[start SB playing in auto init mode]

quit=0;
buffer_flag=0;
while(!quit)
  {
/* double buffered disk read for SB */
  dma=[read DMA pointer]
  if (dma > 16384) && (buffer_flag==0)
    {
    /* if was lo and now playing hi half, load new low half*/
    [read next 16k to low half of buffer]; buffer_flag=1;
    }
  if (dma < 16384) && (buffer_flag==1)
    {
    /* if was hi and now playing lo half, load new high half*/
    [read next 16k to high half of buffer]; buffer_flag=0;
    }

/* very dumb terminal */
  if(kbhit())
    {
```

```
    ch=getch();
    if(ch==27) quit=1; /* esc quits */
    [ output ch to com port ]
    }
  [if char available at com port] putc(ch);
  }

stop_sb();
remove_sb_interupt();
fclose(file)
farfree(buffer_mem)
}

sb_interupt()
{
[acknoledge SB interupt - sb port commands]
out 20h,20h
}
```

I'm bound to have forgotten something, and have brushed over many things
but this is roughly how it would look. There's lots of other error and
end of file stuff as well.

To record the sample there's a few simple but good programs on simtel
which will do the trick nicely. Actually there might even be a
background TSR program that will play the soundfile in the background so

you can just run your normal terminal program. Or maybe use a standard
windows player and terminal, that might be even easier!

Cheers,

Rob

From zs6awk@global.co.za Thu Jun 20 15:03:14 1996
Received: from lin01.global.co.za (lin01.global.co.za [196.3.164.2]) by tapr.org
(8.7.5/8.7.3/1.9) with ESMTP id PAA29685 for <hfsig@tapr.org>; Thu, 20 Jun 1996
15:03:09 -0500 (CDT)
Received: from anx_116.global.co.za (anx_82.global.co.za [196.3.164.132]) by
lin01.global.co.za (8.7.3/8.7.3) with SMTP id WAA06699 for <hfsig@tapr.org>; Thu,
20 Jun 1996 22:01:01 -0200 (GMT)
Message-Id: <199606210001.WAA06699@lin01.global.co.za>
X-Sender: zs6awk@mail.global.co.za
X-Mailer: Windows Eudora Version 1.4.4
Mime-Version: 1.0
Content-Type: text/plain; charset="us-ascii"
Date: Thu, 20 Jun 1996 22:02:32 +0200
To: hfsig@tapr.org
From: zs6awk@global.co.za (Danie Brynard)
Subject: info on C31 DSK ?

Does anybody have more info on the new TI TMS320C31 DSK ? Like amount of RAM
and release date ? How good is the debugger ie compared to say 56002 OnCE ?

danie zs6awk

From Robert.Glassey@nmp.nokia.com Fri Jun 21 08:59:52 1996
Received: from noknic.nokia.com (noknic.nokia.com [131.228.6.10]) by tapr.org
(8.7.5/8.7.3/1.9) with SMTP id IAA11919 for <hfsig@tapr.org>; Fri, 21 Jun 1996
08:59:49 -0500 (CDT)
From: Robert.Glassey@nmp.nokia.com
Received: from samail01.nmp.nokia.com (samail01.nmp.nokia.com [131.228.240.6]) by
noknic.nokia.com (8.6.9/8.6.9) with ESMTP id QAA22122 for <hfsig@tapr.org>; Fri,
21 Jun 1996 16:59:15 +0300
Received: from by samail01.nmp.nokia.com with SMTP
(1.37.109.16/16.2) id AA098535435; Fri, 21 Jun 1996 16:57:15 +0300
X-Openmail-Hops: 2
Date: Fri, 21 Jun 96 14:56:54 +0100
Message-Id: <H000029201e58799@MHS>
In-Reply-To: <1C9943C0.1858@smtpgty.bwi.wec.com>
Subject: Re: [HFSIG:1239] Re:
Mime-Version: 1.0
To: hfsig@tapr.org
Content-Type: text/plain; charset=ISO-8859-1; name="Re:"
Content-Transfer-Encoding: 7bit

>   Thanks for the code Rob, I appreciate it. Can this code be
>   compiled under Linux using GCC+?
>
>   Tony, KE4ATO

Of course its only pseduo code, but it is intended for a PC under DOS.
Under Linux, you'd be better off using the sound card drivers that
already exist for Linux. Sorry I don't know very much about these, only
that they exist.

PS. I've just notices an error in my pseudo code, I allocated a 64
buffer then only used 32K of it (16K for each half of the bouble buffer)
You only really needed to allocate 32K thus:

```
buffer_mem=faralloc(64*1024l); /* 32 k buffer when page aligned */

buffer=[page alignment of buffer_mem pointer ie to get XX00:0000]
/* this is so the DMA doesn't cross a 64K page boundry */
```

This defines a page aligned 32K buffer in an alocated block of 64K.

It would have worked before, it just allocated twice as much memory as
required.

Cheers,

Rob

From forrerj@ucs.orst.edu Thu Jun 27 11:30:11 1996

Received: from ucs.orst.edu (root@UCS.ORST.EDU [128.193.4.5]) by tapr.org
(8.7.5/8.7.3/1.9) with SMTP id LAA05111 for <HFSIG@TAPR.ORG>; Thu, 27 Jun 1996
11:30:10 -0500 (CDT)
Received: from p04.t0.monrotel.com by ucs.orst.edu;
(5.65v3.2/1.1.8.2/13Mar96-1233PM)
id AA27122; Thu, 27 Jun 1996 09:30:02 -0700
Message-Id: <1.5.4.16.19960627173228.344f05c6@ucs.orst.edu>
X-Sender: forrerj@ucs.orst.edu (Unverified)
X-Mailer: Windows Eudora Light Version 1.5.4 (16)
Mime-Version: 1.0
Content-Type: text/plain; charset="us-ascii"
Date: Thu, 27 Jun 1996 09:32:28 -0800
To: HFSIG@tapr.org
From: Johan Forrer <forrerj@ucs.orst.edu>
Subject: Question on orthoganal tone spacing

Hi folks,

HFSIG been quiet lately?

I have a question that I would appreciate some insight - I'm a bit confused.

My multi-channel modulation scheme uses four parallel tones, each modulated at a rate of 75 symbols per second (bauds). If I understood our earlier discussion, each tone should fit into a 2*75=150 Hz slot. Why does this not follow the orthogonal 1/T rule for orthogonal spacing, that would put tones at 75 Hz spacing, not 150 as above?

To check this out, I looked at the MIL-STD-188 16-tone specifications which is quite similar, i.e., 75 baud with 110 Hz spacing. I tried this spacing but experience a substantial amount of inter-channel interference. Demodulation still works but with a serious performance hit - the recovered channel constellations are really smeared. However, when I go to the 150 (i.e. 2*T) spacing - the constallations are text-book quality.

So - what am I missing ?

---------------FYI---
A little more of where this is heading: four 75 baud tones with DQPSK modulation gives 600 bps. Various forms of diversity is possible. The highest form of diversity is when each channel carries the same information in DBPSK, in which case the rate is only 75 bps, but at four times diversity.

It would be possible to do what the Pactor-II folks do and up the DQPSK (4-DPSK) to 8-DQPSK that would essentially yield 1200 bps. Again, it is possible to use various forms of diversity. For example, using 8-DQPSK, and a factor of two diversity, would yield 600 bps.

The idea is to use an interleaved convolutional code, something along the lines that Phil have been working on. The waveform is 600 Hz wide at the -50 dB level, and could possibly be squeezed be into the 500 Hz constraint. However, I do not recommend folks getting ideas of using narrow crystal

filters - severe phase distortion at the filter edges.

If time allows, I hope to do a show and tell demo of the project at the HFSIG meeting during the upcoming DCC (September 20-22, Seattle).

I sincerely hope that this effort will lead to a future open-architecture HF-digital protocol. All specifications will be freely available for those to implement it. Commercial implementations would be encouraged and I would be happy to offer consulting services if anyone is interested. HFSIG is there to encourage development and exchange ideas.

--Johan, KC7WW

From forrerj@ucs.orst.edu Thu Jun 27 13:55:36 1996
Received: from ucs.orst.edu (root@UCS.ORST.EDU [128.193.4.5]) by tapr.org
(8.7.5/8.7.3/1.9) with SMTP id NAA11266 for <HFSIG@TAPR.ORG>; Thu, 27 Jun 1996
13:55:33 -0500 (CDT)
Received: from p02.t0.monrotel.com by ucs.orst.edu;
(5.65v3.2/1.1.8.2/13Mar96-1233PM)
id AA23538; Thu, 27 Jun 1996 11:55:25 -0700
Message-Id: <1.5.4.16.19960627195749.0d3f5f2c@ucs.orst.edu>
X-Sender: forrerj@ucs.orst.edu
X-Mailer: Windows Eudora Light Version 1.5.4 (16)
Mime-Version: 1.0
Content-Type: text/plain; charset="us-ascii"
Date: Thu, 27 Jun 1996 11:57:49 -0800
To: HFSIG@tapr.org
From: Johan Forrer <forrerj@ucs.orst.edu>
Subject: Help communicate/translate in Italian

Hi,

Is there a kind soul out there that can help us communicate with a very keen and interested Italian gentleman?

I have been contacted several times by this keen individual, and although I can help myself in several languages, Italian unfortunately is not one of them - I hate to dissapoint, but I need a bit more than a few hand-waving gestures explaining how to get started using a DSP modem.

The gentleman is located in Milan. Any help would be greatly appreciated.

Thanks a lot.

--Johan

From bm@lynx.ve3jf.ampr.org Fri Jun 28 10:30:07 1996
Received: from lynx.ve3jf.ampr.org (lynx.ve3jf.ampr.org [44.135.96.100]) by
tapr.org (8.7.5/8.7.3/1.9) with SMTP id KAA29585 for <hfsig@tapr.org>; Fri, 28 Jun
1996 10:30:00 -0500 (CDT)
Received: (from bm@localhost) by lynx.ve3jf.ampr.org (8.6.12/8.6.12) id PAA01611

for hfsig@tapr.org; Fri, 28 Jun 1996 15:31:17 GMT
From: Barry McLarnon VE3JF <bm@lynx.ve3jf.ampr.org>
Message-Id: <199606281531.PAA01611@lynx.ve3jf.ampr.org>
Subject: Re: [HFSIG:1243] Question on orthoganal tone spacing
To: hfsig@tapr.org
Date: Fri, 28 Jun 1996 15:31:17 +0000 (GMT)
In-Reply-To: <1.5.4.16.19960627173228.344f05c6@ucs.orst.edu> from "Johan Forrer" at Jun 27, 96 11:31:51 am
X-Mailer: ELM [version 2.4 PL23]
Content-Type: text

> I have a question that I would appreciate some insight - I'm a bit confused.
>
> My multi-channel modulation scheme uses four parallel tones, each modulated
> at a rate of 75 symbols per second (bauds). If I understood our earlier
> discussion, each tone should fit into a 2*75=150 Hz slot. Why does this not
> follow the orthogonal 1/T rule for orthogonal spacing, that would put tones
> at 75 Hz spacing, not 150 as above?

The 1/T rule does apply here - it *should* work with 75 Hz spacing.

> To check this out, I looked at the MIL-STD-188 16-tone specifications which
> is quite similar, i.e., 75 baud with 110 Hz spacing. I tried this spacing
> but experience a substantial amount of inter-channel interference.
> Demodulation still works but with a serious performance hit - the recovered
> channel constellations are really smeared. However, when I go to the 150
> (i.e. 2*T) spacing - the constallations are text-book quality.
>
> So - what am I missing ?

In the case of MIL-STD-188, there is a "guard interval" of about 4.2 ms
at the beginning of the 13.3 ms symbol interval to reduce ISI caused
by multipath. The integrate-and-dump (or equivalent) doesn't start
until after the guard interval, so T = 9.1 ms and 1/T = 110 Hz.

If your scheme isn't working at 1/T spacing, it seems like an
implementation issue. Perhaps you're using some kind of baseband pulse
shaping which is destroying the orthogonality of the tones at minimum
spacing?

Barry

--
Barry McLarnon VE3JF/VA3TCP | Internet: bm@hydra.carleton.ca
Ottawa Amateur Radio Club | AMPRnet: bm@lynx.ve3jf.ampr.org
Packet Working Group | Web: http://hydra.carleton.ca

From forrerj@ucs.orst.edu Fri Jun 28 22:30:35 1996
Received: from ucs.orst.edu (root@UCS.ORST.EDU [128.193.4.5]) by tapr.org
(8.7.5/8.7.3/1.9) with SMTP id WAA28033 for <hfsig@tapr.org>; Fri, 28 Jun 1996
22:30:31 -0500 (CDT)
Received: from p09.t0.monrotel.com by ucs.orst.edu;
(5.65v3.2/1.1.8.2/13Mar96-1233PM)

id AA11728; Fri, 28 Jun 1996 20:30:24 -0700
Message-Id: <1.5.4.16.19960629043259.3667fa2c@ucs.orst.edu>
X-Sender: forrerj@ucs.orst.edu (Unverified)
X-Mailer: Windows Eudora Light Version 1.5.4 (16)
Mime-Version: 1.0
Content-Type: text/plain; charset="us-ascii"
Date: Fri, 28 Jun 1996 20:32:59 -0800
To: hfsig@tapr.org
From: Johan Forrer <forrerj@ucs.orst.edu>
Subject: Re: [HFSIG:1245] Re: Question on orthoganal tone spacing

Hi Barry,

>
>The 1/T rule does apply here - it *should* work with 75 Hz spacing.
>
>> To check this out, I looked at the MIL-STD-188 16-tone specifications which
>> is quite similar, i.e., 75 baud with 110 Hz spacing. I tried this spacing
>> but experience a substantial amount of inter-channel interference.
>> Demodulation still works but with a serious performance hit - the recovered
>> channel constellations are really smeared. However, when I go to the 150
>> (i.e. 2*T) spacing - the constallations are text-book quality.
>>
>> So - what am I missing ?
>
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>at the beginning of the 13.3 ms symbol interval to reduce ISI caused
>by multipath. The integrate-and-dump (or equivalent) doesn't start
>until after the guard interval, so T = 9.1 ms and 1/T = 110 Hz.
>
>If your scheme isn't working at 1/T spacing, it seems like an
>implementation issue. Perhaps you're using some kind of baseband pulse
>shaping which is destroying the orthogonality of the tones at minimum
>spacing?
>
>Barry
>
>--
> Barry McLarnon VE3JF/VA3TCP | Internet: bm@hydra.carleton.ca
> Ottawa Amateur Radio Club | AMPRnet: bm@lynx.ve3jf.ampr.org
> Packet Working Group | Web: http://hydra.carleton.ca
>
>

Thanks for the response - you make a good point. It may well be an
implementation issue as there are many factors involved.

So ... I did a bit of research to understand things a bit better:

I created a signal like the MIL-STD-188 16-tone.
The way that this is done is to modulate all the carriers, sum them in one
signal and then time-domain filter the ensamble using a pulse shape that has

a "rounded" edges. The edges of this pulse shape are made up of 1/2 Hamming rolloff cones that has a duration, the length of the guard zone (2.12 ms). The main portion of the pulse top then is at unity.

The resultant signal audibly sounds reasonable, none of those ugly clicking noises, a kind of a warbling-buzzing sound. My spectrum analyzer showed the carriers are contained as expected and the noise floor is some 30-40 dB down from the tone carriers.

Result: yes, indeed, the demodulator demodulates the individual signals OK, but the recovered constellations are *not* as clean as what I would like to see. Maybe some more implementation issues that may make some improvement?

I compared this to my individually-RC pulse-shaped signal waveform. Note that here I use 2*T channel spacing, which is just slighty more than the MIL-STD-188 tone spacing
(450 vs 440 Hz). I noted significant differences; The dynamic range between constellation points are at least an order of magnitude better, and the spurious levels in the spectrum are at least twice a low.

Thus I suspect that the RC pulse shaping has brought about a cleaner emission and also improved the eye opening substantially. I would be as brave as to say that I have high hopes that the scheme may be usable at 8-DQPSK.

OK so where does this leave us with the issue of orthogonality?

I do suspect that it may be practically impossible to achieve that, i.e., having that accuracy between the tone set. It seem to take only a very small error to distroy the orthogonality to the point where it becomes nearly useless. This may be why the MIL-STD-188 set also does not use the 1/T spacing?. From your description above, the effectively space the tones is 110 Hz spacing (based on the 9.09 ms pulse). So we are already removed from the 1/T orthogonal spacing.

Long story, thanks for your patience, but thought it was worth looking into.

73's

--Johan

From karn@baa01075.slip.digex.net Sat Jun 29 01:32:17 1996
Received: from baa01075.slip.digex.net (root@baa01075.slip.digex.net [204.91.208.66]) by tapr.org (8.7.5/8.7.3/1.9) with ESMTP id BAA11832 for <hfsig@tapr.org>; Sat, 29 Jun 1996 01:32:09 -0500 (CDT)
Received: (from karn@localhost) by baa01075.slip.digex.net (8.7.4/8.7.3) id BAA00279; Sat, 29 Jun 1996 01:25:41 -0400 (EDT)
Date: Sat, 29 Jun 1996 01:25:41 -0400 (EDT)
From: Phil Karn Jr <karn@baa01075.slip.digex.net>
Message-Id: <199606290525.BAA00279@baa01075.slip.digex.net>
To: hfsig@tapr.org
In-reply-to: <1.5.4.16.19960629043259.3667fa2c@ucs.orst.edu> (message from Johan Forrer on Fri, 28 Jun 1996 22:36:04 -0500 (CDT))
Subject: Re: [HFSIG:1246] Re: Question on orthoganal tone spacing

Reply-To: karn@qualcomm.com

>OK so where does this leave us with the issue of orthogonality?

Johan,

I'm out of town so I don't have access to my textbooks, but I *think*
the issue is whether you're using coherent or noncoherent detection.
With a coherent carrier reference at the receiver, you can use carrier
spacing twice as dense as with noncoherent detection.

Since you are using DQPSK, I assume you're doing noncoherent detection.
That could explain your need for greater tone spacing.

Phil

From bm@lynx.ve3jf.ampr.org Sat Jun 29 11:32:55 1996
Received: from lynx.ve3jf.ampr.org (lynx.ve3jf.ampr.org [44.135.96.100]) by
tapr.org (8.7.5/8.7.3/1.9) with SMTP id LAA01365 for <hfsig@tapr.org>; Sat, 29 Jun
1996 11:32:51 -0500 (CDT)
Received: (from bm@localhost) by lynx.ve3jf.ampr.org (8.6.12/8.6.12) id QAA02843
for hfsig@tapr.org; Sat, 29 Jun 1996 16:34:04 GMT
From: Barry McLarnon VE3JF <bm@lynx.ve3jf.ampr.org>
Message-Id: <199606291634.QAA02843@lynx.ve3jf.ampr.org>
Subject: Re: [HFSIG:1247] Re: Question on orthoganal tone spacing
To: hfsig@tapr.org
Date: Sat, 29 Jun 1996 16:34:04 +0000 (GMT)
In-Reply-To: <199606290525.BAA00279@baa01075.slip.digex.net> from "Phil Karn Jr"
at Jun 29, 96 01:44:39 am
X-Mailer: ELM [version 2.4 PL23]
Content-Type: text

> >OK so where does this leave us with the issue of orthogonality?
>
> Johan,
>
> I'm out of town so I don't have access to my textbooks, but I *think*
> the issue is whether you're using coherent or noncoherent detection.
> With a coherent carrier reference at the receiver, you can use carrier
> spacing twice as dense as with noncoherent detection.
>
> Since you are using DQPSK, I assume you're doing noncoherent detection.
> That could explain your need for greater tone spacing.
>
> Phil

This is true, but with coherent detection you can achieve a minimum
orthogonal carrier spacing of 1/(2T). With noncoherent detection, the
minimum spacing is 1/T as previously stated. A good example of a
practical system using 1/T carrier spacing and noncoherent detection
(DQPSK) is the Eureka 147 Digital Audio Broadcast system. For Mode 2
DAB, which we use here in Canada at L Band, the parameters are:

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Transmitted symbol length T = 312.5 us, composed of:
     Tg = 62.5 us (guard interval)
     Ts = 250 us (symbol length in the demodulator)

Carrier spacing = 1/Ts = 4 KHz

There are 384 carriers, producing an overall bandwidth of 1.536 MHz.

I suspect that Johan's modem may not be performing well with 1/T carrier
spacing because he is using windowing which destroys the orthogonality.
Have you tried using rectangular windowing, Johan?  And, of course, if
you implement a guard time to avoid ISI, you must adjust the carrier
spacing so that the orthogonality is maintained in the demodulator
(spacing of 1/Ts rather than 1/T).

Barry

--
  Barry McLarnon  VE3JF/VA3TCP   |  Internet: bm@hydra.carleton.ca
  Ottawa Amateur Radio Club      |  AMPRnet:  bm@lynx.ve3jf.ampr.org
  Packet Working Group           |  Web:      http://hydra.carleton.ca
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